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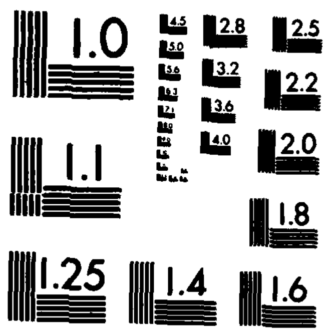

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Human Engineer's Guide to Auditory Displays, Vol. 2: ELEMENTS OF SIGNAL RECEPTION AND RESOLUTION AFFECTING AUDITORY DISPLAYS

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This work reviews the areas of monaural and binaural signal detection, auditory discrimination and localization, and reaction times to acoustic signals. The review was written from the perspective of human engineering and focuses primarily on auditory processing of information contained in acoustic signals. The impetus for this effort was to establish a data base to be utilized in the design and evaluation of acoustic displays. The organization of the reviews presented here is in the form of questions and answers. The questions are numbered and listed in Appendix I to facilitate

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SECURITY CLASSIFICATION OF THIS PAGE (When Data Entered) reference to appropriate sections of the text. Appendix I also contains citations of the scientific literature on which was based the answers to each question. There are nineteen questions and answers, and more than two hundred citations contained in the list of references given in Appendix II. This is one of two related works, the other of which reviewed the literature in the areas of auditory attention, recognition memory, and auditory perception of patterns, pitch, and loudness.

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ABSTRACT

ELEMENTS OF SIGNAL RECEPTION AND RESOLUTION AFFECTING AUDITORY DISPLAYS

This work reviews the areas of monaural and binaural signal detection, auditory discrimination and localization, and reaction times to acoustic signals. The review was written from the perspective of human engineering and focuses primarily on auditory processing of information contained in acoustic signals. The impetus for this effort was to establish a data base to be utilized in the design and evaluation of acoustic displays. The organization of the reviews presented here is in the form of questions and answers. The questions are numbered and listed in Appendix I to facilitate reference to appropriate sections of the text. Appendix I also contains citations of the scientific literature on which was based the answers to each question. There are nineteen questions and answers, and more than two hundred citations contained in the list of references given in Appendix II. This is one of two related works, the other of which reviewed the literature in the areas of auditory attention, recognition memory, and auditory perception of patterns, pitch, and loudness.

ELEMENTS OF SIGNAL RECEPTION AND RESOLUTION AFFECTING AUDITORY DISPLAYS

This work is one of two in-depth reviews of the scientific literature pertaining to the major areas of auditory processing in humans. The areas covered in the reviews presented here are monaural and binaural signal detection, discrimination, and factors affecting reaction times to acoustic signals. More than two hundred scientific reports, reviews, and books provided the information substrate on which this work was based. These documents were selected from the general literature available in the above areas on the basis of their pertinence to achieving a broad understanding of the factors that underlie processing of auditory signals such that it may be applicable to the design and evaluation of non-speech acoustic displays.

As a means of providing the reader with an application-oriented guide to the information presented herein, each unit of information is organized in the form of an answer to a specific question, where the questions themselves were formulated from the perspective of human engineering. There are nineteen such questions of varying degrees of specificity with answers of correspondingly variable lengths and detail. The particular literature on which each answer is based is cited in standard scientific style and listed in full reference format in Appendix II. Each question is numbered throughout the text. These numbers correspond to a listing of the questions in Appendix I which may be used as an index to facilitate the location of questions of interest within the text. The question list in Appendix I also contains, for each question, citations of the pertinent literature to permit direct referral to the list of references in Appendix II.

Although the implications for acoustic displays of much of the material discussed will be obvious, the reader is cautioned against drawing "hard and fast" conclusions for specific applications. Such conclusions may be isolated, out of context, and consequently inappropriate. A general model of auditory processing needs to be developed taking into account the information contained in this volume, as well as its sister volume which covers the areas of auditory attention, recognition-memory, and auditory perception of patterns, pitch, and loudness.[1] Such a model would integrate this information and provide a means of establishing boundary conditions on the applicability of the conclusions drawn from it.

[1]Mulligan, B. E., Goodman, L. S., McBride, D. K., Mitchell, T. M., Crosby, T. N., Gleisner, D. P., Stewart, K. D., and Hitchcock, L., Human Engineer's Guide to Auditory Displays, Vol. 1: Elements of Perception and Memory Affecting Auditory Displays, September 1982.

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1. What factors determine the detectability of monaural signals?

Recognizing that the normal auditory system is, under natural conditions, binaural rather than monaural, this question might best be rephrased to ask, "what factors determine the detectability of diotic signals?" Various studies have shown that the detectability of monaural and diotic signals in noise is equivalent (e.g., Blodgett, et al., 1958; Hirsh and Burgeat, 1958; Mulligan et al., 1967; Durlach, 1972). This is not to say that these two types of signals are effectively the same in all respects. Monaural signals presented through headphones are lateralized completely on the side of the stimulated ear, whereas diotic signals produce auditory images that are located centrally in the vicinity of the midpoint of the binaural axis. Furthermore, monaural signals are not judged to be as loud as diotic binaural signals of the same amplitude. Why, then, is the detectability of these two types of signals effectively the same?

Monaural detection is typically studied under conditions where the input to the non-signal ear is of relatively low magnitude (usually, the only sound present is noise generated under the headphone cushion by bloodflow through vessels) and uncorrelated with the noise presented through the headphone at the signal ear. The effect is a functional isolation of the two ears such that binaural interactions are rendered negligible. Likewise, under diotic (as opposed to dichotic) conditions, binaural interactions are nullified by presenting to the two ears in-phase signals of the same amplitude and frequency against noise backgrounds of the same spectrum level and waveform (interaural correlation of +1). Under either of these two conditions, the signal-contingent interaural imbalances that contribute to the superiority of dichotic binaural detection (see question no. 2) are virtually eliminated.

Except in the case of a severe pathology involving one ear, purely monaural hearing probably cannot occur under natural conditions. However, the relevance of monaural detection, as it is studied under artificial laboratory conditions, is not only that it mimics diotic detection, but that it permits the study of auditory processes (e.g., frequency selectivity, intensity resolution, etc.) independently of binaural complexities. It is also applicable in the case of acoustic communication systems that deliver single-channel signals through headphones.

The major categories of variables that have been shown to influence monaural detection are signal-to-noise ratio, signal frequency, masker frequency or spectrum, signal duration, masker duration, auditory bandwidth, temporal and frequency proximity of signal and noise. Temporal fluctuations in the spectra of complex signals such as speech may also influence detection, as may attentional factors. It may be assumed that the latter did not affect the results of the experiments to be reviewed here.

It is commonly said that the absolute threshold is a masked threshold since, even under "quiet" listening conditions, noise is present in the external auditory meatus (Shaw and Piercy, 1962). However, absolute and masked thresholds do not vary in the same way as a function of signal frequency (Hawkins and Stevens, 1950). The absolute threshold decreases as signal frequency _f_ increases up to about 2 kHz above which it rises as _f_ continues to increase (see Licklider, 1951, for a summary of absolute sensitivity functions). The masked threshold, (or S/N necessary for 75% correct detection, as in the masking study by Green et al., 1959) however, changes less rapidly as a function of _f_, except at low noise levels near absolute threshold. It also is apparent that, for a given signal frequency, as the noise spectrum level increases, the signal level required to reach the masked threshold must increase by approximately the same amount. That is, the signal-to-noise ratio (S/N) is approximately constant for a given _f_ and a specified level of detection performance (in this case, the masked threshold; but note, as discussed below, that performance level increases as a function of S/N). Due to its association with Fletcher's (1940) assumptions regarding the critical band, the masked threshold S/N is often referred to as the critical ratio (Zwicker et al., 1957).

Davis and Krantz (1964) compiled data from a number of studies from which they determined the "minimum audible pressure" at absolute threshold for a range of signal frequencies between 125 Hz and 8 kHz. It shows that the sound pressure level required for detection increases dramatically as _f_ decreases below about 1 kHz (threshold is about 40 dB higher at 125 Hz than it is at 1 kHz). As _f_ increases above 1 kHz, however, only slight increases in threshold levels are needed for detection out to 8 kHz. However, more rapid increases in thresholds above 2 kHz were reported in the older literature summarized by Licklider (1951). In any case, it appears that the auditory system is most sensitive in the frequency region between about 1 and 2 kHz. It seems remarkable that, under ideal conditions, the displacement of the tympanic membrane needed for threshold level stimulation in this frequency region is on the order of the diameter of a hydrogen molecule (Stevens and Davis, 1938). If the sound pressure level is sufficiently great, young listeners can detect tones at frequencies of only a few Hertz (Corso, 1958) or at frequencies as great as 24 kHz (Corso, 1967). Normal listeners can detect sounds delivered to the cochlea through bone conduction (vibration of the mastoid process) at higher frequencies than they can sounds delivered through air to the tympanic membrane (Corso, 1963).

The minimum sound pressure required for threshold detection may depend not only on signal frequency, but also the duration of the signal. Threshold intensity of either tonal signals or noise signals has been shown to decrease as a function of the duration of the signal up to times between 0.05 and 1.0 sec (e.g., Garner, 1947a,b; Garner and Miller, 1947; Green et al., 1957; Plomp and Bouman, 1959; Green, 1960; Zwislocki, 1960, 1969; Olsen and Carhart, 1966). Although various theoretical suggestions have been offered to account for auditory temporal integration, at present the process is neither empirically nor theoretically firm in all respects (see loudness summation in question 8). It seems to be well established, however, that either masked or absolute thresholds may be reduced by increasing the duration of brief signals. Thresholds for tones may be reduced by more than 25 dB (depending on signal frequency) by increasing signal duration from about 1 msec to 1 sec (Plomp and Bouman, 1959). For very brief tonal signals this reduction in threshold with increasing signal duration may be due to a progressively greater concentration of the signal's energy about its central frequency, an effect which may be demonstrated either physically or mathematically. As pointed out by various researchers (e.g., Garner, 1947b; Green et al., 1957; Plomp and Bouman, 1959; Scharf, 1961), if the spectrum of a very brief signal is spread out beyond the width of the critical band centered on the frequency of the signal (i.e., mismatched auditory filter width and signal duration), there may be an energy loss and, consequently, a higher threshold. However, this account probably is not applicable to signals of durations much greater than about 10 msec (depending on signal frequency and assuming a relatively gradual rise-decay) since their spectra are more narrow than the bandwidths reported for the auditory filter. The progressive decrease in threshold that occurs with increases in signal duration to times as great as about 1 sec may, therefore, represent a genuine temporal integration of energy. It seems likely that this integration is neural rather than acoustical (Zwislocki, 1969).

Temporal integration requires some capacity to "hold" the effects of stimulation as they build up over some time. The result is a decrease in threshold. The reverse process, i.e., decay of the after-effects of stimulation, also results in threshold changes, but in the opposite direction. That is, the threshold of a signal may be raised by the after-effects of a sound which immediately precedes it. Various names have been given to this result of stimulation on subsequent detection, e.g., "after-effect masking," "residual masking," "auditory fatigue," "poststimulatory fatigue," etc. Although none of these labels seem to be in all respects accurate discriptors, the one most generally used is auditory fatigue. It should be distinguished from "auditory adaptation" (or "perstimulatory fatigue") which refers to a decrease in loudness of an on-going sound as a function of its duration and intensity. Presumably, both fatigue and adaptation are indicative of reversible reductions in auditory sensitivity as seen in thresholds. In this regard, it is of interest to note that an adapted ear exhibits elevated thresholds (Selters, 1964).

Although there remain areas of uncertainty in the details of auditory fatigue, the general nature of the major relationships is fairly well established. The level of either tonal or noise signals required for threshold detection increases as a function of the level and duration of the pre-exposing sound, and decreases as a function of (1) the time between termination of the exposing sound and onset of the signal (Δt), and (2) the difference in frequency between the exposing sound and the signal (Δf). The effect of Δf, of course, is obtained only in the case of tones or narrow bands of noise. It should be noted that the form of these relationships seems to hold only over limited ranges of exposure intensity, e.g., low, moderate, and high intensities (Elliot and Fraser, 1970). For present purposes it will be adequate to treat separately fatigue effects obtained above and below an exposure level of about 80 dB SL.

The influence of the decaying after-effect of a moderately intense sound exposure on the threshold of a second sound depends on the magnitude of the initial effect and the proximity in time and frequency of the signal to it. At moderate levels of exposure, the magnitude of the effect grows very gradually as a function of the exposure duration, only negligible effects occurring at exposure frequencies below about 500 Hz (Causse and Chavasse, 1947). Since the time required for complete recovery after exposure appears to be independent of the magnitude of the exposure effect, larger initial magnitudes are associated with more rapidly declining decay (recovery) functions. Decay of auditory fatigue occurs as a linear function of log-time (Plomp, 1964a). Decay functions that are linear over time also have been reported (e.g., Luscher and Zwislocki, 1949).

The relationships depicting the decay of auditory fatigue may be regarded as monaural temporal discrimination functions. Indeed, Plomp's listeners were instructed to detect the Δt between two successive noise pulses. The more intense the two pulses were, the smaller was the just-discriminable temporal gap. The smallest gap always occurred with pulses of equal intensity. Reductions in the relative intensity of the second pulse resulted in larger values of Δt. These results are equivalent to auditory fatigue assuming that discrimination of Δt is dependent upon resolution of the two successive pulses and that this resolution is proportional to the magnitude of the declining residual (I_R) of the first pulse, i.e., the intensity difference limen ($\Delta I/I_R$). As intensity increases up to about 20 dB SL, Δt decreases rapidly. Above 20 dB SL, further change in Δt is slight. This is the relationship that would be expected on the basis of intensity discrimination data (see question 7) if Δt is proportional to $\Delta I/I_R$.

The specificity of auditory fatigue has been examined by exposing the ear to a tone of a given frequency and then determining threshold elevation at other frequencies after some fixed time Δt. It appears that the resulting distributions relating threshold shifts to signal frequency are heavily dependent on the intensity of the exposing sound. At low intensities of exposure (below about 50 dB SL for tones), relatively small and symmetrically distributed threshold shifts occur (Causse and Chavasse, 1947). The maximum threshold shift is found at the frequency (f_0) of the exposing tone and falls off symmetrically on either side of f_0 as Δf increases. At higher intensities of exposure the magnitudes of the resulting shifts are greater, as are the widths of the distributions, but the shapes of the distributions are asymmetrical (Munson and Gardner, 1950). For an exposure frequency of 1 kHz at 70 dB SL, Munson and Gardner found that the distribution peaks at f_0, but declines gradually and irregularly as signal frequency increases above f_0. This irregular and asymmetrical spread of the fatigue effect to frequencies above f_0 was found to be even more pronounced a for 100 dB SL exposure. The peak of this distribution shifted up by half an octave above f_0. Others have reported much the same findings (e.g., Davis et al., 1950; Hood, 1950; Zwislocki and Pirodda, 1952; Hirsh and Bilger, 1955). It is likely that the asymmetrical spread of the fatigue effect to, and the emergence of peaks at, frequencies greater than f_0 are the results of mechanical nonlinearities in the cochlear response during intense tonal stimulation. Similar effects have been obtained in simultaneous masking of tones by tones (Wegel and Lane, 1924; Egan and Hake, 1950).

Unlike the relatively short-lived threshold shifts induced by levels of exposure below about 80 dB SL, the shifts due to more intense exposures may endure for minutes, hours, or months. Also, the duration of intense exposures interacts multiplicatively with intensity to a greater the extent than in the case of low or moderate levels of exposure. The longer the duration (T) of the exposure at levels above about 80 dB SL, the greater the magnitude of threshold shift and the longer the recovery time (Mills et al., 1970; Mosko et al., 1970; Ward, 1963, 1970; Ward et al., 1958). The magnitude of the fatigue effect has been found to increase linearly as a function of log T at a rate that is proportional to the intensity above some constant value (Ward et al., 1958). Recovery functions following intense exposures are non monotonic. After an initial decline in threshold during the first minute after termination of the exposure, the direction of the function reverses and peaks at approximately $\Delta t = 2$ min (the "bounce" effect) from which it again declines (Hirsh and Ward, 1952; Hirsh and Bilger, 1955). From $\Delta t = 2$ min, recovery is a linear function of log Δt (Ward et al., 1958).

Both pitch and loudness of post-exposure sounds are affected. The frequencies of tones matched to the pitches of intense exposing tones exceed f_o (Davis et al., 1950), just as do the peaks of the post-exposure threshold-frequency distributions. That is, the pitch of the exposing tone is raised above the pitch normally associated with f_o. In addition, the elevated thresholds due to fatigue are associated with "loudness recruitment," i.e., steeper slopes of loudness functions (Riach et al., 1962; Hickling, 1967) similar to those obtained in cases of permanent hearing loss.

The frequency region of greatest susceptibility to sound induced threshold shifts appears to lie between 2 and 6 kHz (Ward, 1963), however there are large individual differences in these frequency regions. In general, it appears that men are more susceptible to low frequency noise exposures and women are more susceptible to high frequency exposures (Ward, 1966).

Forward masking is obtained under conditions that are operationally equivalent with those in the auditory fatigue experiment (e.g., compare the paradigm employed by Elliot, 1962a, with those of Lüscher and Zwislocki, 1949, and Gardner, 1947). A brief masker is presented to one ear, followed by a silent period Δt, and then the signal of duration t is presented. The smaller the value of Δt (or $\Delta t + t$) for a given level of the masker, the greater the elevation of the signal level that is required for minimal detection (Zwislocki, 1968; Stein, 1960; Elliot, 1962a,b). The reverse condition, backward masking (Elliot, 1962a,b; Raab, 1961; Osman and Raab, 1963; Robinson and Pollack, 1971; Wright, 1964), produces an effect which appears to be similarly dependent on Δt. In this case, the signal is presented first and, as Δt decreases, the level of the signal required for minimal detection increases depending on the masker level. Apparently, the masker disrupts the processing of the signal memory trace if it follows signal offset closely enough. It is possible also that backward masking may be partially due to an elevation of the listener's detection criterion by the masker. In backward masking, the listener must make a detection response after the masker offset rather than after the signal offset. In any case, it is evident that proximity of signal and masker in time raises the signal level required to achieve threshold.

Under conditions of simultaneous masking the signal and masker are physically mixed, i.e., they are present at the same time in the form of a complex wave. Thus detection cannot be improved by moving the signal away from the noise in time (i.e., by increasing Δt), although detection may be improved if the signal and masker are moved apart in frequency. This is possible because the auditory system is frequency selective. If a narrow band noise with a single-tuned power spectrum is the masker, detection of a tonal signal located at the center frequency of the noise (the point of peak power, f_0) will require a larger S/N ratio than if the frequency of the signal is either above or below the region of peak noise power (Egan and Hake, 1950; Scharf, 1971). As the noise power drops off on either side of f_0, so too does the signal power that is required for minimal detection.

The masking of tones by a tone of fixed frequency and intensity was originally investigated by Wegel and Lane (1924). On re-examining this matter, Egan and Hake (1950) found that the effect of tonal masking is similar to that obtained with a noise masker, i.e., threshold elevation of the signal increases as proximity to the masker frequency increases, but with a prominent difference. At the more intense levels of the tonal masker, the masking effect is asymmetrical and irregular on the high-frequency side of the masker. Dips in the function reported by Egan and Hake for a 400 Hz masker at 80 dB are evident at the masker frequency and multiples of it, i.e., 800 Hz, 1200 Hz, and 1600 Hz. The dips are due to interactions between the masker (or its harmonics) and the signal when the frequency difference between the two is small enough for the occurrence of audible combination tones (or "beats" when the signal frequency is very close to either the masking frequency or that of one of its harmonics). Audible combination tones have been located at the difference frequency (f_2-f_1), the cubic difference frequency ($2f_1-f_2$), and several other frequencies ($f_1-n[f_2-f_1]$) by Goldstein (1967). The most prominent combination tone was found to be the cubic. These results were confirmed by Hall (1972).

The asymmetry of tonal masking functions due to the presence of harmonics of the masker probably accounts for the common observation that low-frequency tones mask tones higher in frequency more readily than the converse. But "remote masking" (the name applied by Bilger and Hirsh, 1956, to masking effects located at distances from the masker somewhat greater than a critical bandwidth) also may be produced on the low-frequency side of the masker (Deatherage et al., 1957). The range of frequencies over which these remote effects occur appears to be determined by the intensity of the masker (Greenwood, 1961a); as masker intensity increases beyond about 50 dB above absolute threshold, both the magnitude and breadth of remote masking effects increase. Evidence for this may be seen also in Figure 1-7(b). These effects appear to be the result of nonlinear (and asymmetrical) mechanical responses within the cochlea, i.e., aural distortion. The presence of harmonic distortion products in the cat cochlea was demonstrated by Wever and Bray (1938) and low-frequency distortions by high-frequency stimuli were found in the guinea pig cochlea by Deatherage et al. (1957).

In the case of noise maskers, it is clear that, at any signal frequency, the signal power (and duration) needed to achieve a given level of detection is proportional to the noise power present in the immediate vicinity of the signal. In other words, noise outside of a critical region on either side of the signal frequency (the "critical band") has no effect on the level of the signal needed for detection.

Fletcher (1940) was the first to demonstrate and account for this effect. Brief tonal signals were presented to his listeners against a background of "white noise" and masked thresholds were determined. Then the noise bandwidth was narrowed and the procedure was repeated. Thus, for each signal frequency, Fletcher obtained masked thresholds over a range of noise bandwidths (in each case, the band of masking noise was centered on the signal frequency). He found that reductions in the noise bandwidth had no effect on masked thresholds until the noise was narrowed to a certain critical width. Beyond this point, masked thresholds decreased (signal power required for threshold detection decreased) in direct proportion to further reductions in noise bandwidth. Fletcher's data also indicated that the critical width of the masker increased as a function of signal frequency (see question 6 and Table VI-1). For example, only noise within about ± 20 Hz of 250 Hz exerted any influence on a signal at that frequency, while at signal frequencies of 500 Hz, 1,000 Hz, 2,000 Hz, and 4,000 Hz the critical bandwidths were about ± 20 Hz, ± 30 Hz, ± 50 Hz, and ± 75 Hz respectively. This increase in critical bandwidth accounts for the increase in S/N at the masked threshold as signal frequency increases. However, the detectability of narrow band noise signals against a wide band masker has been shown to be independent of noise signal center frequency if overall bandwidth power and duration are constant (Green, 1960). Further discussion of the critical band may be found in question 6.

Comparisons of S/N values at the masked threshold for tonal signals of various frequencies are based on the assumption that the functions relating detection performance and S/N are parallel across signal frequency. Otherwise, performance level and S/N would be confounded. Green et al. (1959) obtained psychometric masking functions (performance *vs*. S/N) for a range of tonal signals from 250 Hz to 4 kHz and found them all to conform to a cumulative Gaussian function. Even mulit-tone compound signals yielded functions that conformed to those of the single-tone signals (Green, 1958; Green et al., 1959). Since psychometric functions are parallel across frequencies, the particular form of the function is important only in so far as it relates to the theoretical considerations that underlie detection performance in a particular psychophysical task. Hopefully, such theoretical considerations would provide a bridge between the psychometric functions obtained with different psychophysical procedures. For example, psychometric functions generated from an envelope detection model (Mulligan and Elrod, 1970b) were found to describe the relationships between detection performance (pulsed tones; frequencies from 500 Hz to 4 kHz) and S/N obtained by means of two psychological methods. In one case, the proportion correct detections P(C) obtained in a two-interval forced-choice experiment were plotted against S/N, as were those of Green et al. (1959). In the second case, detection performance was determined in a rating scale experiment where listeners generated receiver-operating characteristics (ROCs) for each value of S/N (other parameters held constant) from which the performance measure $\underline{d}_s$ was calculated and plotted as a function of S/N for five signal frequencies (Mulligan et al., 1968). Incidentally, the P(C) and $\underline{d}_s$ measures are related by the fact that P(C) is the proportion of area under the ROC (Green and Swets, 1966).

Psychometric functions from Mulligan and Elrod (1970a) are given in Table I-1 for signal frequencies from 250 to 4000 Hz. The ratios of sound pressures for signals and noise $(S/N_0)^{1/2}$ that are required to achieve percent correct detection between 55% and 95% are tabulated for each signal frequency. Note that, as signal frequency increases, the value of S/N necessary for a given performance level also increases. As Fletcher discovered, this is due to the increase in width of the critical band that occurs with increments in signal frequency.

It is clear that psychometric functions for monaural detection of tonal signals in noise are parallel across signal frequencies at least to 4 kHz and that the increase in S/N with increasing signal frequency may be determined at any level of performance within the working limits of psychometric functions. Alternatively, it may be said that the change in S/N required to bring about a given change in detection performance in a particular psychophysical task is independent of signal frequency. For the two-interval forced-choice task, an increase in $(S/N_0)^{1/2}$ by about 4.2 should raise detection performance from P(C)=65% to P(C)=85%, without regard for signal frequency (see Table I-1). Approximately the same changes in S/N and P(C) would be expected for tonal signals of uncertain frequency (Green, 1961), signals composed of multiple tone components (Green, 1958; Green, et al., 1959), and noise signals of various bandwidths and center frequencies (Green, 1960).

TABLE I-1. $(S/N_0)^{1/2}$ FOR LEVELS OF DETECTION PERFORMANCE [P(C) in 2IFC TASK]

Signal	55%	60%	65%	70%	75%	80%	85%	90%	95%
250 Hz	2.96	4.42	5.61	6.67	7.70	8.73	9.78	11.02	12.71
500 Hz	3.34	4.80	5.99	7.06	8.08	9.11	10.16	11.40	13.09
1000 Hz	4.18	5.64	6.84	7.90	8.92	9.95	11.00	12.24	13.94
1500 Hz	5.08	6.54	7.74	8.80	9.82	10.85	11.90	13.14	14.48
2000 Hz	6.02	7.48	8.67	9.73	10.76	11.79	12.84	14.08	15.77
2500 Hz	6.99	8.44	9.64	10.70	11.73	12.75	13.80	15.04	16.74
3000 Hz	7.97	9.43	10.63	11.69	12.71	13.74	14.79	16.03	17.73
3500 Hz	8.98	10.44	11.63	12.69	13.72	14.75	15.80	17.04	18.73
4000 Hz	10.00	11.46	12.65	13.72	14.74	15.77	16.82	18.06	19.75
Increments For Noise and Tone Compounds	1.46	1.20	1.06	1.02	1.03	1.05	1.24	1.70	

Based on table from Mulligan and Elrod (1970a)

2. What factors determine the detectability of binaural signals?

Detection of binaural signals in noise may exceed monaural detection of the same signals by 20 dB, or more. The factors responsible for this superiority of the binaural system are essentially those on which auditory localization depends, i.e., interaural time or phase, and intensity differences. The interaural imbalances that may be produced by the occurrence of signals in noise at both ears are far more detectable than amplitude increments alone produced by the occurrence of monotic or diotic signals in noise.

The relative difference in the detectability of signals in noise under binaural and monaural conditions traditionally has been evaluated in terms of the MLD or "Masking Level Difference." The MLD is, simply, the difference in decibels between the signal-to-noise (S/N) ratios required to achieve a given level of detectability under binaural and monaural (or diotic) conditions. For example, assume that a S/N ratio of 18 dB is necessary to attain 75 percent correct detection performance when the signal is a 500 Hz tone briefly added to noise at one ear alone. Now, if a duplicate of the noise is also presented to the other ear such that the interaural correlation of the two noises is +1, then we would find that a S/N ratio of only about 10 dB was needed in order to achieve the same level of detectability as when the same signal was added to the noise at one ear only. In this case, the MLD would be 8 dB. Merely by adding +1 correlated noise at the non-signal ear, the S/N ratio required for 75 percent detection performance was reduced from the monaural case by 8 dB. This amounts to more than a 6-fold reduction in signal power as compared with the monaural condition. How can this occur? With correlated noise present at both ears, the addition of the signal at one ear produces an interaural phase shift that contributes powerfully to the detectability of the signal. This does not occur if the noise is presented only in the signal ear.

Taking the example a step further, if two 500 Hz signals are added in phase and at equal amplitudes to the +1 correlated noises at the two ears, the S/N ratio required for 75 percent correct detection performance be again would 18 dB; just what it was in the purely monaural condition! In this case, the detectability of two signals is far worse than just one. It would seem that two signals should be more detectable than one, but they are not because no interaural differences result from addition of this diotic pair of signals to the noise at the two ears. The MLD will be 0 dB even though this is a binaural condition. If, however, these same two signals are added to the noise 180° out of phase relative to each other, a large interaural phase shift would occur and the necessary S/N ratio would be nearly 0 dB. The MLD would be about 18 dB, the equivalent of a 63-fold reduction in signal power from that required for equally detectable monotic or diotic signals.

Licklider (1948) introduced the terms "homophasic," "antiphasic," and "heterophasic" as category names for the various interaural phase conditions. Homophasic conditions are those in which signal and noise are either both in phase or both out of phase, i.e., conditions in which no interaural phase differences arise from either signals or noise.

TABLE II-1. INTERAURAL TEMPORAL RELATIONS BETWEEN SIGNALS AND NOISE

Nm: Noise monotic.

NO: Noise diotic ($\alpha = 1$; $\theta = 0^\circ$) or dichotic ($\alpha \neq 1$; $\theta = 0^\circ$).

Nπ: Noise dichotic ($\alpha = 1$ or $\alpha \neq 1$; $\theta = 180^\circ$).

N : Noise dichotic ($\alpha = 1$ or $\alpha \neq 1$; > 0) or diotic ($\alpha = 1$; $= 0$).

Np: Noise dichotic ($\alpha = 1$ or $\alpha \neq 1$; $p < +1$) or diotic ($\alpha = 1$; $p = +1$).

Nu: Noise dichotic ($\alpha = 1$ or $\alpha \neq 1$; $p = 0$).

- -

Sm: Signal monotic.

SO: Signal diotic ($\alpha = 1$; $\theta = 0^\circ$) or dichotic ($\alpha \neq 1$; $\theta = 0^\circ$).

Sθ: Signal dichotic ($\alpha = 1$ or $\alpha \neq 1$; $\theta > 0^\circ$).

Sπ: Signal dichotic ($\alpha = 1$ or $\alpha \neq 1$; $\theta = 180^\circ$).

Sp: Signal dichotic ($\alpha = 1$ or $\alpha \neq 1$; $p < +1$) or diotic ($\alpha = 1$; $p = +1$) noise.

- -

α: Interaural intensity ratio.

θ: Interaural phase difference.

: Interaural time delay.

p: Normalized interaural correlation coefficient.

- -

Typical combinations of the above include: NmSm, NOSm, NOSO, NOSπ, NOSθ, NpSO, etc.

Antiphasic conditions are those in which either the signals are out of phase and the noise is in phase, or, in which the noise is out of phase and the signals are in phase. Lastly, heterophasic conditions are those in which the noise is independent, or uncorrelated, at the two ears. A more elaborate notation system has evolved (Hirsh and Webster, 1949; Webster, 1951; Jeffress et al., 1956) as a means of indicating the interaural temporal relationships of signals and noise. This system is listed in Table II-1 with several additions to reflect conditions examined in more recent research.

The hierarchy of MLDs for a number of the interaural masking conditions has been determined (Hirsh, 1948; Blodgett et al., 1958). They are, in order of increasing MLD: NmSm, $N\pi S\pi$, NOSO, $NuS\pi$, NuSO, $N\pi Sm$, NOSm, $N\pi SO$, $NOS\pi$ (where NO and SO are diotic, i.e., $\alpha = 1$ and $\theta = 0^\circ$; see Table II-1). It appears to be the case that the conditions NmSm, $N\pi S\pi$, and NOSO are equivalent at 0 MLD (Blodgett et al., 1958; Hirsh and Burgeat, 1958; Mulligan et al., 1967; Durlach, 1972) since no binaural advantage occurs. One or the other of these conditions may thus be taken as the reference for determining the MLD, usually either NmSm or NOSO.

The effect of interaural phase on the masking of signals by noise was first reported by Licklider (1948) and Hirsh (1948). Licklider found that the intelligibility of speech masked by white noise can be altered dramatically by reversing the interaural phase of either the speech signals or the noise, or by varying the interaural correlation of the noise. He found that most masking occurs under homophasic conditions, least masking under antiphasic conditions, and intermediate levels under heterophasic conditions.

Hirsh (1948), using pure tones instead of speech signals, supported and extended Licklider's findings. He found that reversing the phase of both signal and noise ($N\pi S\pi$) had little effect on the masked threshold.

However, reversing the interaural phase of the signal only ($NOS\pi$), or noise only ($N\pi SO$), resulted in large reductions in the masked threshold. The MLD at 100 Hz was small, but increased greatly at 200 Hz. The MLD was largest at 200 Hz (13.6 dB for $NOS\pi$, 7.6 dB for $N\pi SO$), and decreased monotonically as frequency increased to 500 Hz, 1,000 Hz, 2,000 Hz, and 5,000 Hz. Only a slight MLD remained at 5,000 Hz. Similar results were obtained by Webster (1951) for the $NOS\pi$ condition at a number of frequencies from 80 Hz to 6,600 Hz. The largest MLD was found at 250 Hz (20 dB), falling off on either side to about 13 dB at 80 Hz and 4 dB at 6,600 Hz. Hirsh and Burgeat (1958) also obtained pure tone thresholds at several phase and frequency conditions. MLDs of 12.6 dB for $NOS\pi$ and 7.3 dB for $N\pi SO$ were obtained at 250 Hz and MLDs steadily declined as frequency of the signal was increased up to 3,000 Hz. MLDs for the NOSO and $N\pi S\pi$ conditions were nearly identical. The same dependence of MLD on frequency was obtained for the NOSm condition, although the magnitudes were lower; approximately 9 dB at 200 Hz (Hirsh, 1948; Hirsh and Burgeat, 1958). This relationship of MLD to signal frequency for the antiphasic $NOS\pi$ condition has been confirmed more recently by Durlach (1963), Rabiner et al. (1966), Schenkel (1964), and Mulligan (in preparation). It is thus clear that the enhancement of detection of tonal signals in noise under antiphasic conditions shows much the same dependency on signal frequency as does the interaural phase JND and the minimum audible angle (Mills, 1960, 1972; see questions 3 and 4).

The dependence of the MLD on signal frequency also has been examined in studies in which interaural noise correlation was the variable of interest. Comparing NuSm to N0Sm, MLDs of 10 dB, 7 dB and 4 dB were found at 250, 225 and 200 Hz, respectively (Whitmore and Wilbanks, 1965b). In a similar study, MLD values were 9 dB at 250 Hz and 500 Hz, and 4 dB at 1,200 Hz. Only slight MLDs were obtained for 3,000 Hz and 4,000 Hz (Whitmore and Wilbanks, 1965a). It is worth noting that Whitmore and Wilbanks used the uncorrelated noise condition NuSm as their reference for computing MLDs suggesting that NuSm is little, if any, better than NmSm. In fact, Mulligan and Wilbanks (1965) reported a negative MLD of -3 dB for NuSm. Similarly, Rabiner et al. (1966) found relatively small MLDs (4-5 dB) that seemed to be independent of signal frequency (160-1,000 Hz) for the NuS0 and NuS_π conditions.

Wilbanks and Whitmore (1968) also systematically varied interaural noise correlation (p) and found that the MLD increased monotonically from about 1 to 8 dB as p increased from 0 to +1 for the condition $N_\rho Sm$. Virtually identical results were obtained by Dolan and Robinson (1967) for the same condition. Apparently, the magnitude of the interaural phase shift produced by a monotic signal increases as a function of the correlation between the noises at the two ears. Of course, as p approaches +1, the N0Sm condition is approximated.

Robinson and Jeffress (1963) also varied interaural noise correlation for the conditions $N_\rho S0$ and $N_\rho S_\pi$. They found a systematic decrease in MLD from about 12 to 0 dB as correlation increased from -1 to +1 for the $N_\rho S0$ condition, and the opposite trend for the $N_\rho S_\pi$ condition. Thus, in-phase signals produce large MLDs if the correlation of the noise at the two ears is high and negative (as in the $N_\pi S0$ condition), and reversed-phase signals produce large MLDs if the noise correlation is high and positive (as in the $N0S_\pi$ condition).

The relationship between MLD and interaural phase of the binaural signal has also been investigated parametrically. Exploring the $N0S_\theta$ and $N_\pi S_\theta$ conditions with a signal frequency of 200 Hz, Hirsh (1948) varied the interaural phase angle of the signal from 0° to 180° in steps of 30° and measured the masked threshold. With noise diotic ($\alpha = 1$; $\theta = 0°$), MLDs increased as signal phase angle increased: 4 dB at 30°, 7.4 dB at 60°, 10.2 dB at 90°, 12.4 dB at 120°, 11.2 dB at 150° and 13.7 dB at 180°. However, with the noise in reversed phase ($N_\pi S_\theta$), MLDs decreased as the interaural signal phase angle increased, the mirror image of the $N0S_\theta$ trend.

Jeffress et al. (1952) obtained threshold measures while varying the interaural phase angles of both a 500 Hz signal and a wide-band masking noise. The phase of the noise was altered by shifting the 500 Hz masking component. Signal and noise phase angles of 0°, 36°, 72°, 108°, 144° and 180° were examined. The variable of interest was the S/N phase difference. The greatest masking effect was found when the signal and the noise had the same phase position, e.g., $N0S0$, $N_{90}S_{90}$, $N_\pi S_\pi$. The masking effect decreased (MLD increased) as the phase angle between the signal and the 500 Hz component of the noise became more disparate, reaching a minimum when the difference between signal and noise was 180°.

For example, N_{36} S_{144} provided an MLD comparable to that for $NOS\pi$. For the $NOS\theta$ condition, Jeffress et al. obtained results very much like those of Hirsh, i.e., MLDs increased as signal phase angle increased, reaching a maximum of 14 dB at $\theta = 180°$.

Incidentally, these data confirmed early findings (Hirsh, 1948; Licklider, 1948; Hirsh and Webster, 1949) that $NOS\pi$ and $N\pi SO$ conditions do not yield equivalent MLDs. $N\pi SO$ results in slightly less release from masking. The Jeffress et al. MLDs for the $N\theta SO$ conditions did not increase as rapidly with equal increases in noise phase angle (as compared with $NOS\theta$), and the MLD at $N\pi SO$ was just under 13 dB, as compared with 14 dB for $NOS\pi$. These differences may indicate merely that manipulations of interaural phase of sinusoids and random noise components are not equivalent operations.

Jeffress et al. (1962) repeated the experimental procedure of the previous study to investigate the $NOS\theta$ condition for a 167 Hz pure tone. Again, the effect of increasing signal interaural phase angle was to enlarge the MLD, a maximum of 11 dB occurring at $NOS\pi$. This was one of the last studies that employed classical psychophysical procedures to investigate MLDs. After this point, researchers turned to methods derived from the theory of signal detection. These methods require that MLDs be calculated by taking the difference in decibels between two psychometric functions (d' vs. 10 log S/N ratio) at some specified level of detection performance. If the two functions are parallel, the difference between them is constant and the MLD will not vary as a function of performance under a given set of stimulus conditions. If the MLD were to vary in this manner (i.e., if psychometric functions were not parallel), the MLD would be useless as an index of the relative effects of stimulus conditions on masking. Mulligan and Cornelius (1972) examined the question of psychometric invariance and found it to hold on 10 log S/N_0 scales for the conditions NmSm, N0Sm, N0Sπ, NuSm, and NuSπ. More recently, Mulligan (in preparation) has shown that the invariance between monaural and antiphasic conditions holds only if signal frequency is the same for all conditions being compared. Since the slope of the psychometric function varies as a function of signal frequency, MLDs can be determined only from functions involving the same signal frequency.

A 2-alternative forced-choice procedure was employed by Rilling and Jeffress (1965) to determine the 500 Hz signal level required to achieve equal detection (d' = to 1.5) for N0Sφ conditions. The signal phase angles examined were 0°, 15°, 30°, 60°, 120°, 150° and 180°. Colburn and Durlach (1965) employed a 2-alternative forced-choice constant-stimulus method to investigate N0Sφ at phase angles of 0°, 45°, 90°, 135° and 180° for a 500 Hz signal. In the same manner, Durlach (1963) determined MLDs as a function of phase angle for a 200 Hz signal. The MLD values from each of these experiments are compatible with those from previous studies (Hirsh, 1948; Jeffress et al., 1952, 1962). As the interaural phase difference between binaural signals increases relative to that of the noise, the MLD increases nearly linearly up to about 90°. As the phase difference is increased from 90° to 180°, the rate of MLD growth decreases and the function tends to asymptote. The asymptotic levels of these functions decrease as frequency increases above about 200 Hz. The interdependence of signal frequency and interaural phase angle has recently been re-examined by Mulligan (in preparation) and these relationships explored to determine their implications for detection in free sound fields.

Interaural phase differences may be represented as interaural time differences. For pure tones, phase and time are simply related ($\theta = 2\pi f\sigma$) and the relationship of MLD to signal phase angle may be easily transformed to determine the relationship of MLD to signal interaural delay time (σ). From this we would see that maximum MLDs are obtained at time delays equal to 1/2f, where f is the signal frequency. This straight-forward transform does not exist, however, in the case of noise. It has been of interest, therefore, to determine the influence of interaural time delay on the MLD in the condition $N\sigma S0$.

In two related studies, Jeffress et al. (1952, 1962) maintained pure tone signals at 0° interaural phase angle and varied the interaural time delay (σ) of a wide-band masking noise. For 500 Hz, maximum MLDs occurred at time delays of 1 and 3 msec, with the values of 10 and 8 dB respectively. For a 167 Hz signal, the MLD increased to about 4 dB at 1 msec and remained at that level through 3 msec. The 167 Hz data differ in that the maximum MLD is reached much more rapidly and remains at a relatively low value. Langford and Jeffress (1964) extended these studies by introducing even greater time delays in the noise, up to 9 msec. Results were comparable, and showed that for a large interaural time differences, the MLD approximated that which was obtained for an NuS0 condition. Since the signal was in-phase, time delaying the noise at one ear relative to the other ear by small amounts established a phase difference between the noise and signal (interaurally) and thus resulted in an MLD. However, the larger time delay resulted in a decorrelation of the noise.

Using a speech signal delayed in one ear by 470 μsec, Schubert (1956) found that the percent correct speech intelligibility lay between homophasic and antiphasic conditions over five S/N ratios. With longer time delays of 1, 2, 3, 5 and 7 msec, Schubert found no differences between delay conditions. That is, intelligibility did not increase with time delays greater than 1 msec. Similarly, monosyllables that were presented against wide-band masking noise became more intelligible with speech delayed in one ear by 80 msec and 160 msec (Goodnow and Jeffress, 1957).

The improvements in signal detection that occur under antiphasic conditions have been shown to depend on interaural phase, or time, shifts. The detection of signal-contingent phase shifts by the binaural system, however, poses a problem for auditory theory. Since the rate of random fluctuation of the phase and amplitude of gaussian noise increases as a function of its bandwidth, how can the auditory system utilize a signal-induced phase shift (or vector resultant) in wide band noise? Webster (1951) suggested an answer. Due to the narrowness of the auditory bandwidth (critical band) relative to its center frequency, the waveform of the noise affecting a tonal signal changes slowly, i.e., the amplitude and phase of this narrow band process does not change appreciably within the duration of a period of the wave. Hence, the noise outputs from the two monaural auditory filters are slowly varying and sinusoid-like, and the interaural phase shifts produced by the addition of out-of-phase signals are resolvable by the binaural system. This idea was further elaborated by Jeffress et al. (1956) and has since come to be known as the "vector theory." The validity of the theory depends, obviously, on the involvement of the auditory filter in binaural detection. This has been demonstrated in several studies (e.g., Langford and Jeffress, 1964; Bourbon and Jeffress, 1965; Mulligan et al., 1967; Metz et al., 1968; Sondhi and Guttman, 1966; Wightman, 1971; Wightman and Houtgast, 1972).

In addition to interaural time delay and phase effects, interaural amplitude differences may also result in binaural unmasking of signals in noise. Consider the homophasic condition NOSO; noises and signals in-phase and of equal amplitudes at the two ears. The MLD for this condition is 0, i.e., there is no binaural advantage over the purely monaural condition NmSm. However, if the amplitude of one of the signals is reduced relative to that of the signal at the opposite ear by 5 or 6 dB, there occurs a corresponding reduction in masking (increase in MLD) by about 1.5 dB and this trend continues as the amplitude imbalance between the signals at the two ears increases. At the extreme, of course, the condition NOSm (signal at one ear only) is reached and the MLD is about 8 dB (Egan, 1965) or 5 dB (Colburn and Durlach, 1965) for a 500 Hz signal. This trend is reversed if we begin with an antiphasic condition, say NOSπ, in which noises and signals are of equal amplitudes at the two ears, but the two signals are 180° out-of-phase. In this case the MLD is about 14 dB (Egan, 1965) or 11 dB (Colburn and Durlach, 1965) for a 500 Hz signal. Reduction of the relative amplitude of one of the two signals now results in an increase in masking (decrease in MLD) which approaches the value for the NOSm condition as the interaural amplitude difference between signals increases. These same trends were found to occur for the conditions NOS_{90} (decreasing MLD) and NOS_{45} (increasing MLD) by Colburn and Durlach (1965), although the ranges of MLD values were smaller for these two conditions, as would be expected. Thus, if the noises at the two ears are in-phase and of equal amplitudes, and if the signal interaural phase angle is between 180° and 90°, MLD decreases with increases in the interaural amplitude difference between signals. On the other hand, if the signal interaural phase angle is between 45° and 0°, MLD increases with increases in the interaural amplitude difference between signals.

The MLD also may be affected by interaural imbalances in the noise amplitudes at each ear. For example, in the NOSm condition, if the level of the noise in the non-signal ear is reduced below that in the signal ear, the MLD declines proportionately (Blodgett et al., 1962; Mulligan and Wilbanks, 1965; Egan, 1965; Weston and Miller, 1965; Dolan and Robinson, 1967). Likewise, if the noise in the non-signal ear is raised above that in the signal ear, the MLD decreases (Weston and Miller, 1965), although not until the level in the non-signal ear is considerably greater (Mulligan and Wilbanks, 1965). This tendency for the MLD to decline as the noise amplitudes in the two ears depart from equality also has been reported to occur when the noise in the signal ear is raised and lowered above and below that in the other ear (Mulligan and Wilbanks, 1965). A similar effect has been shown to occur in the NOS_π condition when both the signal and noise at one ear are reduced in amplitude while keeping the S/N ratios equal in the two ears (McFadden, 1968).

3. On what does the localization of sound sources in "auditory space" depend, and what are its limits?

For those who are blind, discussions of auditory spatial perception that are limited to man's ability, or lack thereof, to localize sound sources in space must seem drab relative to the richness of their experience of "auditory space." The complex interplay of pitch, tonality, loudness, volume, and density of multiple sounds moving in 3-dimensional space and varying in time and relative significance cannot be adequately addressed from the standpoint of auditory localization, investigations of which have been principally concerned with directional sensitivity and, to a lesser degree, with distance estimation. Unfortunately, the available literature does not provide a scientific foundation adequate to support technical discussion of the more general questions concerning auditory spatial perception. It is therefore necessary for this discussion to be limited to the complex question of auditory localization.

The elemental cues on which directional localization depends are interaural time (or phase) differences and interaural intensity differences. The importance of these cues is discussed elsewhere in this report within the context of binaural time-intensity trading (question 5) and binaural discrimination (question 4). Interaural time and intensity cues arising from sound sources located in space vary in magnitude and effectiveness depending on the frequency (or wavelength) composition of the sound and the angular displacement of the sound source away from the mid-sagittal plane through the listener's head.

The geometry of "auditory space" may be represented in terms of planes and axes through the head, where the latter is regarded as a circular sphere. The three primary planes are (1) the mid-sagittal plane, (2) the mid-transverse plane, and (3) the horizontal plane. The three major axes are (1) the binaural axis, (2) the medial axis, and (3) the longitudinal axis. The axes are oriented at 90° with respect to each other and intersect at a point that is located at the exact center of the spherical representation of the head. The longitudinal axis is a line that runs superior-posteriorly (vertically); the binaural axis is a line that passes through the two ears (laterally); and the medial axis is a line that runs rostral-dorsally (horizontally) through the nose and back of the head. The horizontal plane is positioned at the level of the ears and is precisely quartered by the medial and binaural axes. A circle drawn in the horizontal plane and centered on the cranial sphere intercepts the medial axis at 0° azimuth (directly in front of the face) and 180° azimuth (directly in back of the head). This horizontally oriented circle intercepts the binaural axis at 90° azimuth (on the right side) and 270° azimuth (on the left side). The laterally oriented mid-transverse plane divides the head into front and back and intercepts the horizontal circle at 90° and 270° azimuth. This plane is quartered by the longitudinal and binaural axes. The mid-sagittal plane extends along the medial and longitudinal axes and cuts the cranial sphere into left-right halves. A sound source located at any point on a circle drawn in the mid-sagittal plane, and centered on the cranial sphere, would be equa-distant from the two ears and, consequently, would not (except in the case of high frequency sounds affected by the pinnae and other body parts) result in any interaural differences. The sound source could thus be located directly in front of, in back of, under, or above a listener and appear to originate from the same place if no head movements were permitted. A sound source at any point in space not located in the mid-sagittal plane, however, would be associated with interaural differences, either temporal or intensive. A sound moved in a circle around the head in the horizontal plane would intercept the mid-sagittal plane at only two points, 0° and 180° azimuth, and it would thus be associated with interaural differences at all other angles of azimuth.

Interaural time differences occur because the two ears are not equa-distant from sound sources displaced from the mid-sagittal plane. Assuming the head to be a perfectly circular sphere, Woodworth and Schlosberg (1954, p. 351) showed that this difference in distances to the two ears from a sound source displaced from the medial axis in the horizontal plane could be accurately determined from the following two formulations:

$$(1) \quad \Delta d = k(2\beta)$$

$$(2) \quad \Delta d = k(\beta + \sin \beta)$$

where $\underline{\Delta d}$ is the binaural distance difference, $\underline{k}$ is the radius of the head (8.75 cm) and β is the angle of azimuth, in radians, that locates the sound source on a line that intercepts the center point of the cranial sphere. The two ears are located at 90° and 270° azimuth on the cranial sphere. Equation (1) applies in the case of a source near the head (within one meter), and equation (2) applies in the case of a more distant source. For example, if a source were located close to the head at 30° (0.5236 radians) azimuth, the sound would have to travel about 9.16 cm farther to reach the more distant ear than the ear on the same side as the source. If the source were located at much greater distance from the head, but at the same angle of azimuth, the sound would have to travel about 8.96 cm farther to reach the far ear than the near ear. By comparison, equations (1) and (2) give $\Delta \underline{d}$ values of 27.49 cm and 22.49 cm for near and far sources located at 90° (1.5708 radians) azimuth.

Since sound is propagated through air (at sea level and about 21° C) at approximately 344 m/sec, it will travel one cm in about 0.029 msec. Letting $v = 0.029$, equations (1) and (2) can be written in terms of interaural time difference τ, i.e.,

$$(3) \quad \tau = vk(2\beta)$$

$$(4) \quad \tau = vk(\beta + \sin \beta).$$

From equations (3) and (4) we can now obtain the difference in times of arrival τ at the two ears of a wavefront produced by a sound source located anywhere in the horizontal plane. For example, at 30° azimuth, a source close to the head (1 meter or less) results in a time difference of 0.266 msec and a distant source results in $\tau = 0.260$ msec. At 90° azimuth, near and far sources result in τ values of 0.797 msec and 0.652 msec respectively. It should be noted that the value of τ for a sound source at 90° azimuth is the largest interaural time difference that can be obtained for the average head under free field conditions. In an experiment carried out in an anechoic chamber, Feddersen et al. (1957)

measured interaural time differences as a function of angle of azimuth by inserting small probe microphones into listeners' ear canals and recording the difference in times of arrival of "clicks" at the two microphones. The results were averaged over five listeners and found to agree closely with time differences predicted from equation (4). However, it should be noted that results summarized by Shaw (1974a, b) indicate that interaural time differences for tones may be larger than those predicted by equation (4), even exceeding the limiting case of diffraction theory, i.e.,

$$(5) \quad \tau = vk(3 \sin \beta).$$

Further, it should be noted that Shaw derived his tonal results from phase measurements rather than actual time delays. It is not clear whether this procedure accounts for differences between his results and those of Feddersen et al. (1957), but the latter appear to best fit physical theory (see also Blauert's, 1974, summary).

The following table gives values of τ from equation (4) for increments of 5° in angle of azimuth β from 0° to 90°.

Azimuth (°)	τ MSEC	Azimuth (°)	τ MSEC
0	0.000	50	0.416
5	0.044	55	0.451
10	0.088	60	0.485
15	0.132	65	0.518
20	0.175	70	0.548
25	0.218	75	0.577
30	0.260	80	0.604
35	0.301	85	0.629
40	0.340	90	0.652
45	0.379		

It should be clear that the above function is symmetrical about 90° and 270° ($\pi/2$ and $3\pi/2$ radians) such that, if ϕ is the angle of azimuth in radians then

$$\beta = \phi \quad \text{for} \quad 0 \leq \alpha \leq \pi/2$$

$$\beta = \pi - \phi \quad \text{for} \quad \pi/2 \leq \alpha \leq \pi$$

$$\beta = \phi - \pi \quad \text{for} \quad \pi \leq \alpha \leq 3\pi/2$$

$$\beta = 2\pi - \phi \quad \text{for} \quad 3\pi/2 \leq \alpha \leq 2\pi.$$

The magnitudes of the interaural time differences τ in the above table may seem too small to afford any basis for localization, especially at small angles of azimuth. That this is not the case is evident in the finding that interaural time differences on the order of about 10 μsec are discriminable in the vicinity of the medial axis (Hershkowitz and Durlach, 1969a; Domnitz, 1973) at amplitudes of 40 dB above threshold, or greater. Further, Mills (1960, 1972) showed that just discriminable differences in interaural phase ($\Delta\theta_p$), as measured from $\theta = 0°$ (dichotic tonal signals presented through headphones; Klumpp and Eady, 1956; Zwislocki and Feldman, 1956), were just slightly larger than the interaural phase angles ($\Delta\theta_a$) produced by just discriminable azimuth shifts ("minimum audible angle," $\Delta\phi$) of an actual sound source off the medial axis in the horizontal plane. This close correspondence between $\Delta\theta_p$ and $\Delta\theta_a$ is interpreted to mean that sensitivity to changes in angle of azimuth of sound sources displaced from the medial axis depends on discriminability of interaural phase differences. By extension, it is inferred that interaural phase provides the essential cue for localization of sources of low frequency sounds, below about 1,500 Hz (Hughes, 1940). Mills' results show that the $\Delta\theta_a$ produced by just discriminable shifts in azimuth is a linearly increasing function (with low slope) of log frequency (up to about 800 Hz). As frequency increases above 1 kHz to 1.4 kHz, the rate of acceleration of $\Delta\theta_a$ increases dramatically. For frequencies below 1.5 kHz, values of $\Delta\theta_p$ determined by Zwislocki and Feldman (1956) and Klumpp and Eady (1956) parallel Mills' values of $\Delta\theta_a$, the latter being slightly smaller. It is interesting that the point at which the functions relating $\Delta\theta_p$ and $\Delta\theta_a$ to Log-f begin to depart from linearity corresponds approximately to the frequency (767 Hz) the half-period of which equals the value of τ produced by a source at 90° azimuth (the Hornbostel-Wertheimer constant; Bekesy, 1960, p. 253). This may be mere coincidence.

Although the correspondence between $\Delta\theta_a$ and $\Delta\theta_p$ as a function of frequency has been established only in the region of 0° azimuth, it is reasonable to assume that sensitivity to changes in azimuth ($\Delta\phi$) in other regions of auditory space, e.g., at 60° azimuth, can be accounted for as well in terms of just discriminable interaural phase (or time) differences ($\Delta\theta$). In fact, Mills (1958) showed that the relationship of $\Delta\theta_a$ to frequency forms a family of parallel functions with the reference phase angle θ as the parameter. Increments in θ result in larger values of $\Delta\theta_a$ across all frequencies below 1.5 kHz. Since θ, for a given frequency, increases as a function of ϕ, as does τ (independently of frequency), it is clear that $\Delta\theta_a$ is ultimately dependent on ϕ and τ.

Furthermore, since $\Delta\tau = \Delta\theta/360f$, it is evident that Δ also must increase as a function of τ, as demonstrated by Hershkowitz and Durlach (1969) for $f = 500$ Hz. Whether discrimination of interaural time differences is expressed in terms of $\Delta\theta$ or $\Delta\tau$, as the reference value (θ, τ, or ϕ) becomes large, sensitivity to changes in the temporal cues responsible for auditory spatial resolution diminishes.

The fact that the binaural auditory system can localize (and lateralize) on the basis of interaural phase differences that exist between low frequency signals at the two ears does not mean that phase and time differences are processed as different cues. It does indicate that temporal processing in the binaural system is not limited to initial differences in the times of arrival of a long-duration sound at the two ears. If the two signals are delivered through headphones, interaural phase differences may be varied, even though the two signals are presented simultaneously, and variations in lateralization will occur (Sayers and Cherry, 1957; Sayers, 1964). This is evidence that the binaural system can extract temporal information from waveforms, or microstructures, simultaneously present at the two ears. Such a processing capability seems to be adaptive since the durations of most natural sounds are longer than the differences in their times of arrival at the two ears and these differences are preserved in the temporal relationships between the two ongoing waveforms. Thus, if the signal frequency is sufficiently low (below 1.5 kHz), and provided that the duration is long enough to be processed, binaural resolution of the waveform microstructure can occur. Concerning binaural temporal resolution, several studies have shown that the just discriminable interaural time difference decreases as signal duration increases (Tobias and Zerlin, 1959; McFadden and Sharpley, 1972; Yost, 1974). Recall that interaural time differences of about 10 μsec are discriminable near the medial axis (Hershkowitz and Durlach, 1969a; Domnitz, 1973).

In the case of tonal signals below about 767 Hz, the phase difference across the ears is proportional to the apparent angle of azimuth if the phase or time difference does not exceed τ_h, the Hornbostel-Wertheimer constant (Bekesy, 1960, p. 278). If the interaural time difference of tonal signals between approximately 767 and 1534 Hz exceed $\tau_c = (1/f) - \tau_h$, then temporal information derived from waveform comparisons provides an ambiguous basis for localization. This ambiguity can be avoided only if the difference between the Hornbostel-Wertheimer constant and the period of a tone (1/f) in the range 767 to 1534 Hz is less than the interaural time difference (i.e., $\tau < \tau_c$). Thus, for frequencies below about 767 Hz, τ_h may be taken as the limit of confusion-free zones while, for frequencies between about 767 and 1534 Hz, τ_c is the limit, i.e.,

$$(6) \qquad \tau_c = \begin{cases} (1/f) - \tau_h; \text{ if } \tau_h < (1/f) < 2\,\tau_h \\ \\ 0; \text{ if } \tau_h \geq (1/f) \geq 2\,\tau_h. \end{cases}$$

The interaral phase angles that correspond to these limits are $\theta_h = 2\pi f\tau_h$ for $(1/f) > 2\,\tau_h$, and

$$(7) \qquad \theta_c = \begin{cases} 2\pi(1-f\tau_h); \text{ if } \tau_h < (1/f) < 2\,\tau_h \\ 0; \text{ if } \tau_h \geq (1/f) \geq 2\,\tau_h. \end{cases}$$

For example, the half-period of a 1,000 Hz tone is 0.5 msec. If this tone is swept around the head, the interaural phase difference will increase as the binaural time difference increases. At 5°, 25°, 45°, and 90° azimuth, the phase angles will be 15.8°, 78.5°, 270.2°, and 234.7° respectively. Up to about 62° azimuth the waveform at the right ear will increasingly lead that at the left and the signal source will be localized progressively farther to the right side of the head. However, at about 63° azimuth there is a reversal, i.e., a phase ambiguity. The waveform at the left ear begins to lead and the signal source is localized on the left side at about 62° azimuth. The 1,000 Hz signal is 179.5° out-of-phase at 62° azimuth, but 181.8° out-of-phase at 63° azimuth. If the signal is moved to the most lateral position on the right (90° azimuth), the interaural phase angle will become 234.7°, which is the same as 125.3° with left ear leading, and the singal source will be localized at about 41° azimuth on the left side. This shows that the confusion-free zone for 1000 Hz cannot be greater than about ± 41° azimuth, or ± 125° interaural phase as determined from τ_c in equation (6), i.e., $\theta^\circ = 360f\,\tau_c$.

It is interesting to consider the relationship between angle of azimuth and the frequency of signals at the limit τ_c. At 5°, 25°, 45°, and 90° azimuth, signal frequencies of 1,437 Hz, 1,150 Hz, 970 Hz, and 767 Hz, respectively, are at the limits of their confusion-free zones. In other words, a 1,437 Hz signal can be unambiguously localized only within the narrow range of ± 5° azimuth while the confusion-free zone for a signal of 970 Hz is ± 45° azimuth. No confusion-free zone exists for a signal of 1,534 Hz. This is noteworthy, since the auditory system does not appear to process temporal information in the waveform microstructure of frequencies above about 1,500 Hz. As the magnitude of angular displacement of a sound source increases, the frequency of the signal must decrease to avoid ambiguity. A parallel finding using headphones is that the maximum degree of lateralization of tones increases as frequency decreases.

Recall that, from equation (4), the interaural time difference produced by any signal at 90° azimuth is 0.652 msec, the half-period of a 767 Hz tone (the Hornbostel-Wertheimer constant; von Bekesy, 1960; Hornbostel and Wertheimer, 1920). It is merely coincidental that 0.652 msec is the period of a 1,534 Hz tone, roughly the frequency above which the binaural system fails to process phase information and for which there is no angle of azimuth that does not yield phase ambiguities.

The above discussion pertains primarily to tonal signals lateralizations of which are periodic within interaural time differences equal to their periods. Similar, but non-periodic results have been obtained for noise signals (Blodgett et al., 1956; Pollack and Trittipoe, 1959a,b; Jeffress et al., 1962), where interaural time delay and interaural correlation were the parameters varied. The influence of interaural time delay on the lateralization of "clicks" has also been found to yield comparable results (Hirsh and Sherrick, 1961; Teas, 1962; Guttman, 1962a; Harris et al., 1963; Babkoff and Sutton, 1966), as it has for the lateralization of speech (Cherry and Taylor, 1954).

In all of the studies referred to thus far, interaural time differences were produced either by time-delaying signals presented through headphones, or by displacing a sound source away from the medial axis in the horizontal plane. As stated before, interaural time (or phase) differences arise under free field conditions because of the differing distances a sound must travel to reach the two ears (low frequency, long wavelength sounds diffracting around the head), the distance difference increasing as a function of lateral displacement in the horizontal plane. It is important to realize that these same distance differences may be produced by sources not located in the horizontal plane. For example, all points along the perimeter of a circle parallel to the mid-sagittal plane and centered on the binaural axis (imagine a wheel located several meters to the side of the head on a rod passing through the two ears) will be equa-distant from the near ear and also equa-distant from the far ear (excluding from consideration any interference by body parts). Consequently, the distance difference to the two ears will be the same for any point on such a circle, including the two points at the intercepts of the circle with the horizontal plane. A sound source moved to any point on this circle would, therefore, result in the same interaural time difference.

Essentially the same could be said for interaural intensity differences with two prominent exceptions, viz., binaural distance differences are of negligible importance, and structures of the body and head exert a significant influence, especially at high frequencies. The latter exception arises because the shorter wavelengths of the higher frequency sounds at which significant intensity differences occur are not diffracted around the head (or other body parts) as readily as the long wavelenghts of low frequency sounds. Consequently, much of the energy in the short wavelength sounds tends to be scattered and absorbed by the head and body resulting in an intensity difference at the two ears that is referred to as the "sound shadow." The higher the frequency, the sharper the shadow (i.e., the greater the energy drop across the ears) produced by the head, the pinnae, and other body parts. Obviously, the magnitude of the shadow varies with the directional location of the sound source relative to the orientation of the head, the interaural intensity difference increasing as the angle of azimuth changes from 0° to 90°. However, at a frequency of, say, 15 kHz, the sound shadow produced by a source at about 160° azimuth in the horizontal plane would be different than that produced by a source at 20° azimuth due to the effect of the pinnae (note that the interaural time difference obtained with low frequency sound for each of these locations would be approximately the same). Thus, at high frequencies, interaural intensity differences would not be the same for all points on a laterally positioned circle oriented in the sagittal plane and centered on the binaural axis, even though distance differences would be approximately the same.

It was stated above that binaural distance differences were of negligible importance in accounting for interaural intensity differences. Of course, the pressure of a sound wave diminishes as a function of the distance the wave travels through air due to diffusion (the pressure of spherically spreading waves decreases by 6 dB for each doubling of the distance travelled from the source), molecular absorption, viscosity and heat conduction in the atmosphere. However, it appears that the binaural distance differences are too small to afford any significant decrease in pressure across the ears (for distant sources) relative to that produced through absorption and scattering by the head. Interaural intensity differences are thus almost completely due to the shadowing of the far ear by the head, the effectiveness of which increases as a function of frequency. For example, if a source is located at 90° azimuth, the intensity difference for a 250 Hz tone is about 2 dB; for a 1,000 Hz tone, it is about 6 dB; and for a 10,000 Hz tone, it is about 20 dB (Feddersen et al., 1957; Gulick, 1971, p. 189; Shaw, 1974a,b).

Lateralization studies (e.g., von Bekesy, 1959, 1960; Pinheiro and Tobin, 1969; Flanagan et al., 1964; Guttman, 1962; Moushegian and Jeffress, 1959; Whitworth and Jeffress, 1961; Harris, 1960; Sayers, 1964; Sayers and Lynn, 1968) all show that the binaural auditory image is lateralized toward the ear of greatest intensity, the degree of lateralization increasing as the interaural intensity difference increases. Although the literature is not in agreement on the magnitude of the interaural intensity difference required to achieve a completely lateralized image, most studies indicate that the image is lateralized fully to one side if the intensity difference is approximately 10 dB. An important exception to this was reported by Moushegian and Jeffress (1959). They found that, for a given intensity difference, the degree of lateralization depends on signal frequency. By contrast, von Bekesy (1960) reported that, for a given intensity difference, the degree of lateralization is independent of frequency for tones greater than 3 kHz.

While lateralization of binaural images is strongly affected by intensity imbalances at the two ears, it appears that intensity discriminability is not. Hershkowitz and Durlach (1969a) found that the binaural JND for intensity remained approximately constant at about 0.8 dB even for interaural intensity imbalances as great as 58 dB which approximates the monaural condition. Their signal frequency was 500 Hz. They also found that the binaural intensity JND was nearly constant over a large range of interaural time delays, from 0 to 1,000 μsec. They did find, however, that some slight improvement in the binaural intensity JND occurred (from about 1.5 dB to 0.6 dB) as the intensity of the binaural tones increased from 10 to 78 dB above absolute threshold. It appears, therefore, that binaural intensity discrimination remains highly stable in the face of large interaural temporal and intensive imbalances.

Discriminability of interaural intensity differences (JNDs) as a function of signal frequency was shown by Mills (1960, 1972) to follow an irregular function. The JND increases gradually from about 0.65 dB to 0.7 dB as frequency goes from 250 Hz to 500 Hz, but accelerates more dramatically as frequency continues to increase above 500 Hz up to 900 Hz where the JND peaks at about 1 dB. The peak is not sharply tuned and the function has dropped just slightly at 1 kHz. Between 1 kHz and 1.5 kHz the function drops off almost linearly, the rate of decline diminishing above 1.5 kHz as frequency approaches 2.5 kHz where the function reaches a low point at about 0.5 dB. Between 2.5 kHz and 5 kHz the function again increases gradually to about 0.65 dB and then rolls off in a broad peak showing little decrease up to 6 kHz, but declining at a greater rate between 6 kHz and 10 kHz, finally reaching a value of about 0.45 dB at 10 kHz.

The above function was obtained under listening conditions in which dichotic signals were presented to the two ears through headphones. In order to determine how well the above JND function accounted for the discriminability of azimuth changes ("minimum audible angle") of free field sound sources, Mills determined the actual interaural intensity differences produced by just discriminable displacements (from 0° azimuth) of sources that ranged in output frequency from 250 Hz to 10 kHz. He found that the resulting intensity difference function increased nearly linearly from 0 dB at 250 Hz to about 0.2 dB at 800 Hz. Above 800 Hz, the function positively accelerated to a peak of about 0.60 dB at 1.5 kHz. Between 1.5 kHz and 5 kHz, the intensity difference function followed closely the JND function. However, as signal frequency increased above 5 kHz, the intensity difference function rose sharply reaching a peak at 8 kHz and declining sharply thereafter. The peak value was about 1.7 dB, an intensity difference that exceeded the dichotic intensity JND by more than 1 dB. Why intensity differences exceeding the dichotic JND are required to detect changes in azimuth at frequencies above 5 kHz is not understood. It is clear, however, that detection of azimuth changes at frequencies below 1.5 kHz cannot be accounted for on the basis of the interaural intensity differences produced by changes in azimuth since these differences are smaller than the intensity JNDs. Recall that, for signal frequencies below 1.5 kHz, Mills found that sensitivity to azimuth was attributable to interaural phase (or time) differences. Between 1.5 kHz and 5 kHz, the correspondence between the JND function and the intensity difference function is sufficiently close to warrant the assumption that sensitivity to azimuth is entirely accounted for by interaural differences in intensity.

The above findings were obtained in the vicinity of 0° azimuth. Mills (1958) also determined minimum audible angles as a function of frequency at 30°, 60°, and 75° azimuth. The results of this experiment that were obtained at frequencies below 1.5 kHz were described in our discussion of interaural time differences. As was pointed out before, the minimum audible angle at 30° azimuth is larger than it is at 0° azimuth for all frequencies. Above 1.5 kHz, however, the minimum audible angle for 30° aximuth exceeds that for 0° azimuth by a larger amount (about 4° as compared with about 0.5° below 1.5 kHz; at 5,000 Hz the difference is greatest at about 10°).

Between 1.5 kHz and 2 kHz, minimum audible angles are indeterminately large if the source is located to the side by as much as 45° azimuth. Large, but measurable minimum angles are obtained between 3 kHz and 5 kHz at 60° and 75° azimuths.

Beyond the obvious conclusion that binaural discrimination of angular directions on the basis of interaural intensity differences is poor, if not non-existent, for sources more than 30° to the side, little can be said authoritatively. As a best guess, the increasingly poor directional discrimination that occurs as angle of azimuth increases toward 90° might be understood in terms of the rate at which the interaural intensity difference changes with changes in angle of azimuth. At any frequency capable of producing intensive differences at the two ears, the most rapid change in interaural intensity difference with azimuth occurs in the vicinity of 0° azimuth. Beyond about 30° azimuth, the intensity difference functions for frequencies below about 2 kHz tend to flatten out, showing relatively little change as angle of azimuth increases. As frequency increases, however, the intensity difference functions become more steep, flattening out only in the vicinity of 90° azimuth. The obvious implication is that the minimum audible angle will be smaller in the vicinity of 0° azimuth because a smaller change in the direction of a source will result in a large enough change in the intensive difference to exceed the JND. At 30°, however, larger changes in azimuth (depending on signal frequency) are required to produce changes in the interaural intensity difference that are large enough to be detected.

The role of interaural intensity differences in auditory localization thus appears to be of significance only at frequencies above about 1.5 kHz. Between about 1.5 kHz and 5 kHz, interaural intensity differences completely account for sensitivity to changes in the direction of sound sources away from the medial axis, as do interaural time differences below 1.5 kHz. Smaller directional displacements are discriminable if the source is located in front of the listener than if it is located to one side for both time and intensity cues. However, the degree of lateralization of the binaural image is greater for sources displaced farther to one side since interaural time and intensity differences increase as the source is moved from 0° to 90° azimuth. Of course, lateralization is toward the ear leading in time and receiving the greater intensity (Trimble, 1928).

An early study by Stevens and Newman (1936) investigated accuracy of localization of a free field sound source as a function of signal frequency. Tones of brief duration were generated from a source which was rotated about the listener's head in the horizontal plane (the listener and apparatus was located on the top of a building as a means of reducing reflected sound). For each position of the source, the listener estimated its direction to within 15°. Errors in localization were averaged over source locations for each signal frequency and plotted as a function of frequency. The resulting error function peaks in the vicinity of 3 kHz, the region of poorest localization. At frequencies above and below this region, errors in localization diminished.

Head movements were not permitted in this study and, at frequencies below about 1.5 kHz, the ability of listeners to distinguish between sources located either in front or in back was only a little better than chance. By contrast, errors in front-back discrimination were minimal above about 4 kHz. Between 2 kHz and 4 kHz front-back errors diminished approximately linearly as a function of log-frequency. Again, these findings suggest the operation of intensity cues at high frequencies where wavelengths are sufficiently short to be attenuated by the pinnae such that intensities associated with a source in back of the head are less than those from a source in front.

That front-back discriminations may be attributed to the influence of the pinnae on high frequency sounds was established in an experiment by Batteau et al. (1965). They fitted two silicone molds of pinnae with microphones which were connected by circuitry to a pair of headphones worn by a listener located in a separate sound chamber. The two artificial pinnae were positioned on a rod with the normal space separating them but without a head. On the basis of the acoustic outputs from the microphones inside the artificial pinnae, listeners were able to identify the direction of the source with reasonable accuracy in the horizontal plane and at elevated positions. If the artificial pinnae were removed from the microphones, however, the listeners' judgements became erratic. The sound source was a maraca which produced a broad-band sound. It seems unlikely that any of this sound below about 8 kHz contributed to the accuracy of localization since wavelengths would not be sufficiently short to interact with the pinnae. Furthermore, above 8 kHz the spectrum of the sound probably varied differentially at the two ears as a function of azimuth providing an informationally rich signal for localization. In this connection, it has been shown that small changes in the spectral distribution of sound are detectable (Karlin, 1945).

It is interesting to note that Batteau et al. (1965) also found that listeners could follow the direction of movement of the sound source even when only one artificial pinna and a microphone was used. Again, localization deteriorated when the pinna was removed. This situation is roughly equivalent to monaural localization with head movements permitted. In a study comparing accuracy of monaural and binaural localization, Batteau and Plante (1962) found that azimuth errors increased from $\pm 4^\circ$ to $\pm 30^\circ$ if one ear of the listener was occluded. Head movements were allowed. Monaural localization of unfamiliar and stationary sound sources without head movements probably would not be possible, although this condition does not appear as yet to have been investigated.

Listeners in the studies reviewed thus far indicated the apparent direction of the sound source by such means as pointing, stating the angle of azimuth, or simply indicating "right" or "left." A more direct approach was taken by Sandel et al. (1955). Their listeners "aimed" a noise source in the directions from which they judged tonal signals to emanate. The listeners were seated in an anechoic chamber and operated a remote control device which rotated a speaker (the noise source) located at the end of a boom. Tonal signals were presented from 1 of 3 locations, 0° or $\pm 40^\circ$, and alternated with the noise until the listener indicated that a match had been achieved. Since broad-band noise is more accurately localized than tones, the errors of localization obtained in this study presumably reflect the degree of accuracy achievable in judging the directions of tonal sources.

Sandel et al. found that the average error of localization was small for frequencies between 500 Hz and 1,000 Hz regardless of the location of the source. The largest systematic errors were obtained in the range 1.5 kHz to 3.0 kHz and, above 1.5 kHz, listeners displayed a marked tendency to underestimate the ±40° azimuth sources. Errors of localizing the 0° azimuth source were considerably smaller across all frequencies than for the two lateral sources. The range of errors (scatter) for all three sources was greatest (listeners were least certain) in the frequency range 1.5 kHz to 2.5 kHz. Although Sandel et al. did not require their listeners to make just discriminable azimuth settings, their results conform in all important respects with those of Mills (1958, 1960) As would be expected, therefore, sensitivity to azimuth may be taken as an index of localization accuracy.

In the studies done by Stevens and Newman, Sandel et al., and Mills, listeners were prevented from moving their heads. Of course, this was necessary in order to establish the accuracy of localization with sources positioned at fixed angles relative to the orientation of the head. If listeners are allowed to "search for" a source with head movements, accuracy of directional localization improves. Front-back discrimination with head movements is superior to that achieved without head movements (Burger, 1958; Thurlow and Runge, 1967). However, discrimination of the elevation of sound sources is only minimally aided by head movements (Thurlow and Runge, 1967). This finding is not consistent with expectations based on the analysis of interaural time and intensity cues provided by Wallach (1939, 1940). His analysis showed that the magnitude of change in interaural cues produced by head rotation is greatest if the source is located in the horizontal plane and diminishes as the source is elevated, becoming 0 when the source is directly above the head on the longitudinal axis. Pivotal movements of the head, however, would be expected to result in larger changes in time and intensity cues at the two ears than rotational movements when the source is elevated. Thurlow and Runge found that neither rotational nor pivotal movements contributed significantly to accuracy of localization under such conditions.

Wallach's own experimental results were more in keeping with theoretical expectations. He found that, if the source was moved in the horizontal plane in precise synchrony with rotations of the listener's head, always staying directly in front at 0° azimuth, the source was localized above the head between 90° and 70° elevation. This was consistent with the expected result since, if rotational movements of the head produced no changes in interaural or intensity relations, the source must be located either directly above or below the head. Wallach's listeners tended to localize the source above the horizontal plane rather than below. When the source was displaced laterally in the horizontal plane by an angle of azimuth which remained constant regardless of rotational head movements, listeners localized the source at an elevation that approximately equalled the complement of the constant azimuth angle, a result also to be expected.

Wallach obtained a somewhat surprising result when he arranged the source such that it rotated at twice the angle that the listener moved his head. When the initial position of the source was 0° azimuth, after head rotation listeners described the source as being stationary and located behind them. When the initial position of the source was located at 180° azimuth, listeners reported that the source was stationary and located in front. It is not clear why the source was heard as being stationary, rather than as moving around the head at a rate faster than the rate of head movement. In a further experiment, Wallach set the rate of source movement relative to head movement to a value intermediate between 2:1 and 1:1. Listeners again reported stationary sources with reversals of the initial source position, but, in this case, listeners described the source as being elevated above the horizontal plane. Mills (1972) showed that an elevated source would be predicted from an analysis of the interaural cues for this condition. Again, however, it is unclear why the source was perceived as stationary.

An interesting demonstration of the interplay of head movements and changing interaural cues in determining directional localization was provided by Klensch (1948). He inserted a tube into each ear. The distal end of each tube terminated in a funnel. Both funnels were aimed at a sound source. By advancing one funnel toward the source while moving the other funnel farther away, Klensch was able to manipulate the interaural time and intensity cues independently of head movements. If the left funnel was advanced and the right withdrawn as the listener rotated his head to the right, the source was localized in front of him. However, if the listener rotated his head to the left under these same conditions, the source was localized in back of him. These results served to illustrate what happens under ordinary conditions of listening. If a sound source is located in front of us, a rotation of the head to the right will produce an increase in the intensity and lead time of the sound at the left ear relative to the right ear. The same interaural changes occur with rotation of the head to the left if the source is located in back of the listener. Consequently, it is the direction of head movement that identifies the location of the sound source in this situation.

Directional location of sound sources within enclosed spaces, or areas with reflective surfaces, would be deteriorated if acoustic reflections were as effective as the sounds transmitted directly to the two ears, especially in unfamiliar areas. Fortunately, the influence of echos on localization are suppressed. This is known as the precedence effect. In the case of two successive bursts of sound that reach the ears at slightly different times, the soundburst that arrives first exerts the strongest influence on the direction of localization. For example, given two speakers that are located at 45° and 315° azimuth, and equa-distant from the center of the head, if a burst of sound from the speaker at 45° azimuth precedes a burst from the speaker at 315° azimuth by no more than several msec, the two bursts will be heard as a single sound originating from the direction of the speaker on the right side (Wallach et al., 1949; Steinberg and Snow, 1934). If the soundburst from the speaker at 315° is more intense (or the speaker is closer to the listener) than that from the speaker at 45°, the precedence effect will be partially overridden (depending on the relative intensities of the two sounds) and the single perceived sound will be localized at an intermediate position. If the two sounds are separated in time by more than some upper limit (about 6 msec for clicks) they are heard as separate sounds.

Wallach et al. (1949) examined the precedence effect in some detail under headphone listening conditions where the acoustic stimuli were dichotic pairs of clicks. The first dichotic pair of clicks was followed 2 msec later by a second pair. The delay times of the two successive pairs were antilateral, i.e., if for the first pair the left ear led in time, then for the second pair the right ear led. The interaural time difference between the second pair of clicks was set at a value between 0 and 600 msec and the listener adjusted the interaural time difference between the first pair to a value just sufficient to return the auditory image to the midpoint of the binaural axis. Wallach et al. found that small time differences between members of the first pair were adequate to offset much larger time differences between members of the second pair. In other words, the time differences of the first pair were considerably more effective than those of the second pair in influencing lateralization.

The foregoing discussion focuses on the matter of directional localization of sound sources and the interaural acoustics on which it depends. It may seem odd that so much attention would be given to the directional component of the relative location of sound sources without even a mention of the distance component. Certainly the latter could not be excluded from any meaningful discussion of the localization of objects in visual space. But then, if we mean by localization "pointing at" something sensed, the relative direction and distance of an object in visual space is given the instant it is seen, not before. By contrast, a source in auditory space may be heard, its relative direction ascertained with fair accuracy through head movements, and then its distance estimated only poorly (vis-a-vis vision). The short wavelengths, apparent straight-line propagation, and high velocity of light make it an ideal carrier of distance information (linear perspective, size, interposition, texture, etc.), but the eye already must be aimed in the direction of the source if this information is to be received. Sound within the human range of hearing, on the other hand, consists of relatively long wavelengths, bends around objects, and is propagated at a comparatively slow velocity. While it does not carry the rich distance information contained in reflected light, it does permit the extraction of a high order of directional information from the temporal and intensive differences that develop at the two spacially separated and physically shielded (by the head and pinnae) auditory receivers. Significantly, this system does not have to be "aimed" at the source to locate its direction. But aiming movements of the head appear to be the natural response to sounds originating from outside the visual field, a response which serves to bring the eyes into alignment with the source and thereby permit the pinpoint spatial analysis characteristic of vision.

This is not to say that the auditory system is incapable of distance perception. It does appear, however, that the distance cues available to the binaural system enable no finer resolution of sound source distance than that achievable by the monaural system alone. For example, the curvature of a spherical wavefront produced by a low frequency source changes as a function of distance from the source and this results in interaural phase and intensity differences (for laterally located sources) that also vary as a function of distance. These binaural disparities were originally calculated by Hartley and Fry (1921) and measured by Wightman and Firestone (1930) using an artificial head. They found that measurable interaural phase and intensity differences changed as a function of distance (for a source near 90° azimuth) out to at least 400 cm. Hartley and Fry next carried out an experiment to determine whether listeners could discriminate distances on the basis of these binaural disparities. Negative results were obtained even when listeners were presented with combinations of phase and intensity differences that exceeded any natural condition.

Of the monaural cues for sound source distance, it appears that intensity is the one most utilized. In a free sound field the pressure of a spherically spreading wave decreases by 6 dB for each doubling of the distance it travels from the source due to diffusion and energy absorption in the atmosphere. The intensity or sound pressure level of a sound at the ears, however, is not an unambiguous cue to the distance of its source. The intensive information must be supplemented by sufficient contextural information (intensive, spectral, and temporal) to permit recognition of the source. Otherwise, if sound sources are unfamiliar, listeners are unable to discriminate between the distances of weak sources that are near and strong sources that are far. Coleman (1962) made clear the importance of familiarity for auditory distance estimation, and von Bekesy (1949) showed that the judged distance of familiar sounds (speech) increased as the intensity at the ears decreased.

Not only does the overall intensity of sound diminish as a function of the distance it travels through the atmosphere, but the frequency spectrum of sound also changes. Energy in the higher frequencies of propagated sound is more subject to molecular absorption than that in the lower frequencies. The result is a kind of filtering by the atmosphere which alters the spectral content of sound in proportion to the distance it travels, the energy contained in the higher frequencies being reduced by a relatively greater magnitude. Thus, if a source emits a broad spectrum sound, the relative proportion of high frequency energy will diminish as the distance from the source increases. Again, however, the effectiveness of this cue depends on the listener's familiarity with the sound (Coleman, 1963). For impulsive sounds, von Bekesy (1960) found that listeners estimated the distance to decrease as low frequency content increased. This effect was limited to distances of about 1.2 meters.

An additional cue for distance occurs in spaces that contain reflective surfaces. The intensity at the ears of a sound reflected off a surface, relative to the intensity of the sound which travels directly to the ears, increases as a function of distance. von Bekesy (1960) investigated the effect of this cue on distance estimation. He varied the ratio of the intensity of the reflected (delayed) sound to the intensity of the direct sound and found that listeners perceive the sound as moving away from, or toward, them in the predicted direction as the ratio changed. Of course, the delay of the reflected sound also varies as a function of distance relative to that of the direct sound. Together, the relative delays and intensities of reflected and direct sounds are referred to as reverberation. The kinds of reverberation that will occur in any space is peculiarly determined by the acoustic properties of that space, and cannot be considered an unambigous cue to distance. Consequently, familiarity with sounds heard in reverberant spaces is an important determiner of distance localization. The monaural character of this cue is illustrated in the well-known reverberation effect produced at a microphone in an enclosed room. The closer a speaker is to the microphone, the less the reverberation. A radio listener can thus estimate the distance of a speaker from the microphone more readily in a "live" room than in a heavily damped enclosure.

4. How sensitive is the binaural auditory system to interaural time and intensity amplitude differences?

The binaural resolving capacity of the auditory system is remarkable, especially in the discrimination of interaural time differences. For example, in the frequency region, 500 to 1,000 Hz, interaural time differences of less than 20 μsec can be discriminated (Durlach, 1972). In the case of long duration low frequency noise, binaural timme differences on the order of 6 μsec can be resolved (Tobias and Zerlin, 1959).

Such findings are obtained by presenting stimuli through headphones to each ear. For tonal stimuli, if the variable of interest is the minimally discriminable time difference ($\Delta\tau$), the ratio of amplitudes at the two ears (α) and the reference interaural delay time (τ) are set at one pair of values and the listener attempts to detect a difference between this condition and a second one containing a small change in τ. As the size of this change increases, discrimination performance improves. The magnitude of the change in τ required to reach criterion performance is the difference limen $\Delta\tau$. In this kind of experiment, tones are presented for some duration (e.g., 300 msec) that is sufficiently long for temporal information to be extracted from the waveforms at the two ears. Delay times of tones may be transformed to phase differences by means of the relation $\tau = \theta/2\pi f$, where θ is phase angle in radians, $\underline{f}$ is tonal frequency, and delay time τ is in seconds.

Minimally discriminable interaural time differences ($\Delta\tau$) were determined as a function of frequency by Klumpp and Eady (1956) and Zwislocki and

Feldman (1956). Both studies agree that, for $\tau = 0$, the relationship of Δ to log-f is "V"-shaped and reaches its minimum value of $\Delta\tau$ (approximately 15 μsec) at 1 kHz. For frequencies above 1 kHz, $\Delta\tau$ increases rapidly such that, at about 15 kHz, $\Delta\tau$ becomes indeterminately large, i.e., no temporal discrimination occurs. At frequencies below 1 kHz, $\Delta\tau$ gradually increases as frequency decreases. It is about 18 μsec at 500 Hz, about 30 μsec at 250 Hz, and about 57 μsec at 125 Hz. In terms of phase angle, the smallest discriminable difference ($\Delta\theta$) is approximately 2.3° at 125 Hz. From this point the value of $\Delta\theta$ increases as a positively accelerating function as frequency increases toward 15 kHz, becoming indeterminately large.

The dependence of the minimally discriminable interaural amplitude difference ($\Delta\alpha$) on frequency was investigated by Mills (1960). It was found that $\Delta\alpha$ varied in an irregular fashion between roughly 1 and 0.4 dB in the frequency range 250 Hz to 10 kHz. Not only was there no systematic dependence of $\Delta\alpha$ on f, but these binaural values of $\Delta\alpha$ are of the same order of magnitude as amplitude differences that can just be discriminated monaurally. Apparently, the binaural system provides no advantage in amplitude discriminability over the monaural.

Hershkowitz and Durlach (1969a) examined closely the dependence of Δ and $\Delta\alpha$ on the absolute level (A) at the two ears (where $\alpha = A_l/A_r = 1$), the interaural time delay (τ), and the interaural amplitude ratio (α). Tonal frequency was constant at 500 Hz throughout all conditions. Duration of the tone was 300 msec with a 50 msec rise-decay time.

The relationships of $\Delta\tau$ to A, and $\Delta\alpha$ to A, were determined by setting $\tau =$ 0 and $\alpha = 1$ and then finding the minimal magnitude of change in τ and α required for reliable discrimination (i.e., $\Delta\tau$ and $\Delta\alpha$) at each of a range of values of amplitude between A = 10 and 75 dB SL. It was found that both Δ and $\Delta\alpha$ decreased as amplitude increased. The minimally discriminable time difference ($\Delta\tau$) decreased in a negatively accelerating function and leveled off at approximately 10 μsec at 40 dB SL. Increments in A beyond this level did not result in any further decrease in $\Delta\tau$. The minimally discriminable amplitude difference ($\Delta\alpha$) decreased in a roughly linear trend with a shallow slope from about 1.5 dB at 10 dB SL to about 0.5 dB at 75 dB SL. It is apparent, therefore, that interaural discrimination of differences in both time and amplitude is better if the signals at the two ears are well above the absolute threshold.

The relationships of $\Delta\tau$ to τ, and $\Delta\alpha$ to τ, were determined by setting $\alpha =$ 1 and A = 50 dB SL and then finding the minimal magnitude of change in and α required for discrimination over a range of values of τ between 0 and 1,000 μsec. For instance, if one of the 500 Hz tones presented to the two ears was delayed by 400 μsec on each trial, a change of nearly 20 μsec from the 400 μsec delay was required for discrimination as compared with about 10 μsec ($\Delta\tau$) at a delay (τ) of 0. In this manner, Hershkowitz and Durlach (1969a) found that, as the interaural time delay increased from 0 to 400 μsec (at 500 Hz), the magnitude of the just discriminable increment $\Delta\tau$ increased from about 10 to 20 μsec in an approximately linear trend. In other words, discrimination of interaural time differences is almost twice as good at $\tau = 0$ as it is at $\tau = 400$ μsec for 500 Hz signals. In the case of the minimally discriminably amplitude difference ($\Delta\alpha$), it was found that $\Delta\alpha$ remained approximately constant at about 0.9 dB across all values of τ between 0 and 1,000 μsec. That is, discrimination of interaural amplitude differences is insensitive to interaural time delays.

The relationships of $\Delta\tau$ to α, and $\Delta\alpha$ to α were determimed by setting the level of the tone in the left ear at 50 dB SL and then reducing the level at the right ear. At each value of $\alpha = A_l/A_r$, the magnitudes of $\Delta\tau$ and $\Delta\alpha$ required for discrimination were determined. It was found that as α increased from 0 to 30 dB, the magnitude of the change in interaural time differences that is just discriminable ($\Delta\tau$) increased as a positively accelerating function from about 10 μsec at $\alpha = 0$ to more than 100 μsec at $\alpha = 30$ dB. Clearly, discrimination of interaural time differences is best if the levels at the two ears are equal, but this temporal discrimination is remarkably stable, being only slightly affected by an amplitude imbalance of as much as 20 dB. In the case of the minimally discriminable amplitude difference ($\Delta\alpha$), it was found that $\Delta\alpha$ remained approximately constant at about 0.8 dB as α was varied from 0 to 55 dB, a very large interaural imbalance.

Hershkowitz and Durlachs' findings indicate that sensitivity to changes in interaural time differences is best under the following conditions: (1) at moderate to moderately high amplitudes; (2) when the change in τ to be detected is made from $\tau = 0$ rather than from $\tau > 0$; and (3) when the levels in the two ears differ by no more than 10 to 20 dB. Findings (2) and (3) were confirmed in all important respects by Domnitz (1973). Hershkowitz and Durlachs' findings also show that sensitivity to changes in interaural amplitude differences is better the more intense the two signals, and that neither time delay nor severe interaural amplitude imbalance affects this discriminability.

Since, for small to moderate values of $\Delta\tau$ and $\Delta\alpha$ (where $\tau = 0$, $\alpha = 1$), and levels of A > 6 to 10 dB SL, the ratio $\Delta\ /\Delta\alpha$ is comparable with the time-intensity trading ratio obtained from lateralization experiments (see question no. 5), it is commonly argued that the sensory cue responsible for the $\Delta\tau$ and $\Delta\alpha$ discriminations is a change in position of the auditory image along the phenomenal plane running between the two ears, i.e., a lateralization cue. Whether or not changes in lateralization are responsible for $\Delta\tau$ and $\Delta\alpha$, there can be little doubt that interaural time and intensity differences are necessary for observations of (1) lateralizations of auditory images, and (2) azimuth locations of apparent sound sources. Sensitivity to azimuth should, therefore, be largely accounted for in terms of sensitivity to $\Delta\tau$ and $\Delta\alpha$.

Data supporting this idea was obtained by Mills (1960, 1972) who found that sensitivity to azimuth (minimum audible angle, $\Delta\phi$) can be accounted for in terms of discriminability of interaural time (or phase angle) for frequencies below 1.5 kHz, and in terms of interaural amplitude differences for frequencies between 1.5 kHz and about 5 kHz. For frequencies above 5 kHz, Mills found that sensitivity to azimuth was less than that predicted by discriminability of interaural amplitude differences. It is not clear why the latter result occurs. These data apply only to angles of azimuth with the medial axis as the reference point. That is, an actual source located at a point in space directly in front of the observer was moved laterally to the position at which the observer was just able to detect a change in azimuth. Interaural phase angles ($\Delta\theta_a$) and amplitude differences ($\Delta\alpha_a$) produced at the two ears were measured for each discriminable displacement of the source ($\Delta\phi$). The values of $\Delta\theta_a$ thus obtained paralleled those from headphone experiments, although the former were slightly smaller at each signal frequency. Between 1.5 kHz and 5 kHz the values of $\Delta\alpha_a$ also coincided closely with those obtained from headphone experiments. It therefore appears that discriminability of the angular direction of tonal sound sources can be attributed to the interaural time and intensity differences that occur as a result of diffraction of low frequency sounds around the head and the "sound shadow" created by the head at the higher frequencies. These findings are also consistent with those obtained in experiments on localization (see question 3).

Binaural resolution of interaural time differences depends not only on the reference value of τ, the interaural amplitude ratio α, and the signal frequency, but also on the duration of the signal. Tobias and Zerlin (1959) found that, for low-pass pulsed noise signals, $\Delta\tau$ decreased from 24 to 6 μsec as pulse duration increased from 8 to about 700 msec at which point no further decrease occurred. It appears that 6 μsec is the smallest value of $\Delta\tau$ that has been reported.

5. How do interaural time and intensity differences "trade" in determining the lateralization of sounds?

Under free-field listening conditions, if a single sound source in the horizontal plane were displaced to one side of a listener's head, the ear on that side would lead the opposite ear in time of arrival of the sound. Also, if the sound contained components sufficiently high in frequency to reduce diffraction around the head, the ear oriented toward the sound source would receive a greater intensity of sound. In this case, both interaural time and intensity differences would locate the sound source in the same general direction relative to the orientation of the listener's head in the sound field. Changes in either the location of the sound source or the orientation of the listener's head would result in positively correlated variations in interaural time-intensity differences.

In the case of multiple sound sources, variations in interaural time-intensity differences may not be positively correlated (Sandel et al., 1955). That is, the intensity at one ear may exceed that at the contralateral ear while the latter ear may lead in time. In such a case, these two interaural cues are opposed ("antilateral") and the auditory image may be diffused, or it may be split into several images located at different places along the phenomenal plane that corresponds to a line between the two ears, or it may remain a compact image that is merely shifted by some amount to the right or left of the midpoint of the phenomenal plane between the ears. If the amount by which one ear leads the other in time is precisely counter-balanced by a greater intensity at the other ear, the image may be located at the midpoint. The magnitudes of these opposing interaural time and intensity differences required to counter-balance each other has come to be known as the time-intensity "trade-off."

It appears that the earliest reports on time-intensity trading were due to Hornbostel and Wertheimer (1920), and Klemm (1920). These researchers presented "clicks" to the two ears with a small delay (e.g., 100 msec) imposed between the arrival times of the two clicks in each pair. They found that, within limits, the displacement of the acoustic image toward the leading ear could be counteracted by increasing the relative intensity of the click presented to the non-leading ear. Some value of the ratio of intensities at the two ears could be found for which the displaced image (due to a time difference) would be restored to the midpoint of the plane between the ears.

Shaxby and Gage (1932) studied antilateral time-intensity trading by means of low frequency tones rather than clicks. Instead of varying interaural differences in time of arrival of the leading edges of transient sounds (clicks), Shaxby and Gage varied the interaural phase difference between tones presented through headphones to the two ears. The relationship between interaural phase and time may be stated as $\theta = 2\pi f\tau$, where θ is the phase angle in radians, $\underline{f}$ is the frequency of the two tones, and τ is the time delay between corresponding points on the two waveforms. For example, the period of a 500 Hz tone is 2 msec. If a 500 Hz tone is presented to each ear, one of which leads the other by $\tau = 1$ msec, then the interaural phase angle is $\theta = \pi$ radians, or 180°. Similarly, if $\tau =$ 0.5 msec, then $\theta = 1/\pi$ radians, or 90°. Obviously, if τ is held constant and frequency is changed, θ must also change, e.g., a 0.5 msec delay between 1 kHz tones would be represented by a phase angle of π radians.

Shaxby and Gage determined the value of θ required to counter-balance a displacement of the auditory image ("lateralization") induced by an antilateral intensity difference at the two ears. They found that, for a given interaural intensity difference, as frequency increased so too did the value of the interaural phase angle necessary to restore the image to the midpoint position. By solving for $\tau = \theta/2\pi f$, Shaxby and Gage determined that, regardless of the values of $\underline{f}$, τ remained approximately constant for each interaural intensity difference. That is, in order to counter-balance a given intensity difference, a given time difference was required regardless of the frequency of the two inputs. Their results thus indicated that time-intensity trading is insensitive to frequency. This should not be taken to mean the time-intensity trade can be accomplished by means of interaural phase shifting at any frequency. As shown by Hughes (1940), the binaural system is insensitive to interaural phase differences for frequencies above approximately 1.5 kHz (also see Klump and Eady, 1956; Zwislocki and Feldman, 1956). Of course, this frequency limitation does not apply if the interaural time difference is created by the leading edges of two acoustic transients such as "clicks."

A parametric study of time-intensity trading with click stimuli was conducted by Deatherage and Hirsh (1959). They set the intensity levels of the clicks delivered to the two ears at some difference value and then determined the magnitude of the time delay necessary to restore the auditory image to the midpoint position. Intensity and time differences were antilateral, i.e., the ear opposite the one receiving the click of greatest intensity led in time. The more intense click was set at one of three levels (40, 60, or 80 dB) and the other click was set at some lesser magnitude. Intensity differences ranged between 0 and 30 dB. The amount by which the less intense click had to precede the arrival time of the more intense click to achieve a return of the image to midpoint was found to increase as a function of the magnitude of the intensity difference. For intensity differences larger than about 6 dB, the magnitude of the required time delay increased nearly linearly with increases in intensity difference. Furthermore, at the greater intensity levels (e.g., 80 dB), a given time delay would counter-balance a larger intensity difference. For example, a time delay of 0.5 msec would counter-balance an intensity difference of only about 5.5 dB if the more intense click were 40 dB, but the same delay would counter-balance an intensity difference of about 19 dB if the more intense click were 80 dB. This indicates that the effectiveness of interaural time differences is greater for intense than weak stimuli. Essentially the same conclusion was reached by David et al. (1959) who found that, for intense stimuli, interaural time differences were on the order of 5 times more effective in counter-balancing lateralization displacements due to interaural intensive differences.

Although the above studies were carried out under conditions where stimuli were delivered through headphones, they indicate the importance of temporal cues in the localization of sources of pulsative, or transient, sounds containing high-frequency energy. Cherry and his colleagues (Leakey et al., 1958; Cherry and Sayers, 1959), in the course of their research on binaural "fusion" of sounds of differing frequency content, also concluded that the localization of high frequency transient sounds may depend on differences in the times of arrival of the sharply rising leading edges of such sounds at the two ears. These studies indicated that the effectiveness of interaural time differences between transients depends not only on intensity, but also frequency content.

The influence of frequency content on time-intensity trading was examined directly by Harris (1960). The stimuli were pulses containing energy either below 1 kHz or above 4 kHz. He found that, for low-frequency pulses, a smaller interaural time difference was required to counter-balance a given antilateral intensity difference than was required for high-frequency pulses. The relationship between time delay and intensity difference was approximately linear for both pulse types. Time delay increased at a rate of 25 μsec per decibel intensity difference for low-frequency pulses, and at a rate of 60 μsec per decibel for high-frequency pulses. Time differences at low frequencies were thus found to be more effective in counter-balancing interaural intensity differences than time differences at high frequencies.

Recent investigations of the time-intensity trade have employed signal detection methodology and focused on detectability of differences rather than lateralization. In the latter case, the listener determines the magnitude of an interaural time difference (or intensity ratio) that is required to counteract the lateral displacement of the auditory image induced by an antilateral interaural intensity difference (or time difference). If the time-intensity trade were perfect, the auditory image would be restored precisely to the midpoint position along the phenomenal plane between the two ears. Presumably, this central position of a balanced trade-off is precisely the same as the location of the auditory image under conditions of diotic signal presentation, i.e., when the signals at the two ears arrive simultaneously, in phase, and at identical amplitudes. Consequently, discrimination on the basis of lateralization between the diotic condition and the completely balanced trade-off condition should be nill. The auditory images should be located at the same phenomenal position under both conditions. Furthermore, as time-intensity combinations depart from balance, discriminability between the diotic and unbalanced trade-off condition should improve. This is, essentially, what Hafter and Carrier (1972) found.

Discrimination performance improved as the time-intensity values of tonal pulses (500 Hz tones of 125 msec duration at 70 dB SL) departed from the point of balance. However, discrimination performance at the points of balance in Hafter and Carriers' study increased as the values of time and intensity differences became large. That is, even at the points of balance, the diotic and trade-off conditions were discriminable at large values of interaural time and intensity differences. While the trade-off values of time and intensity taken from Hafter and Carriers' balance points correspond well with those from the lateralization experiments, the discriminability obtained for the larger magnitude trade-offs seems somewhat problematic. Perhaps this was due to split or diffuse auditory images such as those reported in the older literature for large interaural time-intensity differences. Other discrimination studies of the tradeability of interaural time and intensity (e.g., Babkoff et al., 1973; Gilliom and Sorkin, 1972; Hershkowitz and Durlach, 1969b) are in general agreement with these results.

There are several conclusions to be drawn from the data on time-intensity trading that may be important for acoustic displays, especially displays through binaural headphones. For low frequency tones (below 1,500 Hz), the value of the interaural phase angle (θ) needed to balance a given interaural intensity difference increases as a function of frequency. However, this relation is such that the value of the interaural time delay ($\tau = \theta/2\pi f$) is independent of frequency. This means that, regardless of signal frequency, displacements of lateralized tonal images due to given intensity differences at the two ears may be balanced by given antilateral time differences. This frequency independence of the time-intensity relation does not hold, however, for transient sounds, e.g., "clicks." In the case of such sounds, a given time difference will counter-balance a larger intensity difference at low frequencies than at high frequencies.

Generally, the time-intensity trade is such that, as the interaural intensity difference increases, the magnitude of the time difference needed to restore the auditory image to its medial location also increases, nearly linearly for intensity differences greater than about 6 dB. Furthermore, the effectiveness of time differences in offsetting intensity differences improves as the absolute intensities at the two ears increases, i.e., time differences are more effective for intense sounds. Also, if time-intensity values are not large, and if they are completely balanced, the image is not discriminably different from the diotic condition, i.e., the condition in which no interaural time or intensity differences exist between binaural inputs.

6. What are the limits of auditory frequency discrimination?

Since the original work by Knudsen (1923) on the discriminability of frequency differences between tones, the question of frequency resolution in the human auditory system has been examined from several points of view and by means of several psychophysical methods. Regardless of theoretical approach or experimental method, all studies agree that absolute frequency resolution is best at low frequencies (below about 1,000 Hz) and decreases as a function of increasing frequency.

An early study on tonal frequency discriminability that remains, perhaps, the most definitive work on the subject was contributed by Shower and Biddulph (1931). They found that the difference limen (DL) for frequency is dependent upon both frequency and intensity above absolute threshold, i.e., sensation level (SL). At all frequencies, as SL increases, the frequency DL decreases. For example, at a signal frequency of 250 Hz the absolute DL was found to be approximately 9 Hz at 5 dB SL, 5.5 Hz at 10 dB SL, 3.3 Hz at 20 dB SL, 2.8 Hz at 40 dB SL, and 2.4 Hz at 60 dB SL. Thus, it appears that the auditory system differentiates among frequencies more finely, its resolving power is greater, if the intensities of the frequency components are substantially above absolute threshold, i.e., up to about 20 dB SL. Little further improvement in frequency resolution occurs beyond this level.

Shower and Biddulph also found that the absolute value of the frequency difference limen varies as a function of frequency. For a fixed value of SL, as frequency increases up to about 1,000 Hz, the absolute DL remains approximately constant. However, as frequency increases above 1,000 Hz, the absolute frequency difference required for discrimination increases approximately linearly. While this says, for example, that the absolute DL at 4,000 Hz will be about twice as great as the DL at 2,000 Hz (roughly 30 Hz as compared with 16 Hz for an SL of 5 dB), it should be pointed out that, in a relative sense, frequency resolution is actually about the same at 4,000 Hz and 2,000 Hz. That is to say, the ratio of the absolute DL to the base frequency ($(f_c - f_o)/f_o$; Weber fraction or relative DL) is approximately equal in these cases. In fact, there is little improvement in the relative DL with increases in frequency above about 1,000 Hz. Below this frequency, though, as frequency increases the relative DL improves dramatically. For example, for a fixed SL of 5 dB, as frequency increases in the steps 125 Hz, 250 Hz, 500 Hz, 1,000 Hz, 2,000 Hz, and 4,000 Hz, the relative difference limen decreases in the steps 0.0625, 0.0360, 0.0170, 0.0095, 0.0080, 0.0065.

In a sense, the relative frequency DL may be taken as an indication of the efficiency of tuning in the auditory system. By comparison, the absolute frequency DL may be taken as an indication of the acuteness of frequency resolution in the auditory system. Both quantities are indicators of frequency selectivity.

The relationships between DL, SL, and frequency obtained by Shower and Biddulph were, essentially, duplicated by Harris (1952) who employed a different psychophysical procedure. Shower and Biddulphs' listeners detected just noticeable pitch changes in a tone that was "warbled" in frequency, i.e., the tone oscillated in frequency between two values, f_o and f_c, at a fixed rate. Harris' listeners judged whether the second of two successively presented tones was of a "higher" or "lower" pitch than the pitch of the first tone. While Harris' data display the same relationships among the variables DL, loudness level, and frequency as do the data of Shower and Biddulph, Harris' absolute DLs are somewhat smaller and thus may indicate better frequency resolution. It seems likely that this difference in the absolute DL for frequency was the result of methodological differences, if not greater practice received by Harris' listeners who were reported to have been trained observers. It has been shown that the DL for frequency can be reduced through practice (Campbell and Small, 1963).

Utilizing a procedure adapted from signal detection methodology (the two-alternative forced-choice task), Henning (1967) examined the relationship between discrimination performance and signal-to-noise ratio (S/N) at signal frequencies of 250 Hz, 1000 Hz, and 4000 Hz. He found that performance (percent correct discrimination of a constant frequency difference, Δf) improves as S/N increases, reaching asymptotic levels determined by the value of Δf. Larger values of Δf were found to yeild higher performance asymptotes. This improvement in frequency discriminability with increasing S/N is consistent with the findings of Shower and Biddulph (1931), and Harris (1952), assuming that variations in sensation level, loudness level, and signal-to-noise ratio are effectively the same. This is a reasonable assumption if the "quiet" condition for the absolute threshold is actually one in which low-level noise is present (as Shaw and Piercy, 1962, have shown it to be), and if discrimination performance is dependent on S/N rather than absolute values of signal and noise (as Henning, 1967, has shown it to be). Henning (1967) also found that frequency discriminability is not independent of the performance level at which the frequency DL is taken, i.e., larger Δf values are required to achieve higher levels of performance for a constant value of S/N. In a later study, Henning (1973) found that the S/N required to achieve a given level of discrimination performance can be reduced under binaural listening conditions if the signals are presented out-of-phase at the two ears.

Tone duration also may influence the size of the frequency DL if stimuli are brief. According to Henning (1970), the frequency DL decreases as duration increases up to 50 msec for 250 Hz and 1000 Hz tones, but up to only 25 msec for 4000 Hz tones.

In all of the above experiments care was taken to insure that the ear was not stimulated by a complex sound, e.g., the sum of two or more pure tones presented simultaneously. Investigators were interested in determining the maximum resolving power of the auditory system uncorrupted by complex combination effects induced by simultaneously present tones. Other investigators, however, have been interested in determining how acutely the auditory system can differentiate among the frequency components of such complex inputs. While such studies tell us less about the maximally achievable resolving capacity of the auditory system, they may tell us more about the degree of resolution that can be expected in response to realistically complex sounds.

In one such study, Plomp (1964b) presented a complex of two tones to listeners and had them determine which of two probe tones most closely matched either the higher or lower tone in the complex. The probe tones were not "on" at the same time as the complex. Listeners controlled presentation of the three sounds by means of a 3-way switch. The middle position turned on the complex sound, and the right and left positions each turned on one of the two probe tones. The frequency of one probe tone was set equal to one of the components of the complex sound, and the frequency of the other probe tone was set midway between the two components of the complex sound. Listeners had to decide which probe tone matched one of the complex sound components. The variable of interest was the minimum frequency separation between these components which was necessary for listeners to achieve criterion performance in selecting the matching probe tone. The smallest value of the frequency DL obtained by Plomp (1964b) was about 22 Hz at 200 Hz. For frequencies below 200 Hz, the DL increased dramatically indicating that frequency selectivity is relatively poor for low frequency tonal components of complex sounds, at least in discrimination tasks such as that employed by Plomp. For example, at 100 Hz, a DL of nearly 37 Hz was required for a successful match. By comparison, the pure tone resolutions obtained by Shower and Biddulph (1931) and Harris (1952) in this frequency region ranged between approximately 2 Hz and 9 Hz, depending on the effective tonal intensity. Interestingly, at these low frequencies, with separations of complex sound components less than 30 Hz, nearly all of the matches obtained by Plomp corresponded to the intertone setting. These results were confirmed by Nordmark (1978) by means of a slightly different procedure.

Above 200 Hz, Plomp's (1964b) DLs were approximately symmetrical (on log-frequency) with the DLs he obtained at low frequencies. At 100 Hz the DL is about the same as the DL at 500 Hz. As frequency increases beyond

500 Hz, the DL continues to increase such that it is approximately 78 Hz at 1,000 Hz, 175 Hz at 2,000 Hz, and 480 Hz at 4,000 Hz.

Terhardt (1968) also investigated the problem of 2-tone discriminability. His procedure required listeners to decide whether a complex, beating sound consisting of two frequency components could be assigned one, or two, pitches. As the frequency components of the complex sound were moved farther apart, their individual resolution improved. In this case, the DL represents that separation in frequency between the components of the complex at which the listener's judgements changed from 1 to 2 pitches. This would seem to be a somewhat easier task than that employed by Plomp, and, not surprisingly, Terhardt's DLs are slightly smaller than those obtained by Plomp. Above 500 Hz Terhardt's data parallel those of Plomp, but, below 200 Hz the two sets of data diverge. Whereas Plomp's DLs increase as frequency decreases below 200 Hz, Terhardt's continue to decrease reaching a DL of about 18 Hz at 100 Hz.

Plomp and Mimpen (1968) extended the technique employed by Plomp (1964b) to the determination of the frequency separability of the harmonics of complex sounds. As in the earlier study, listeners operated a 3-way switch to select 1 of 2 probe tones that equalled in frequency one of the harmonics of the complex sound. The other probe tone was set at a frequency midway between this harmonic and the next higher, or lower, harmonic. Plomp and Mimpen found that their listeners were able to distinguish no more than 5 to 7 harmonics of a complex sound. Plomp and Mimpen hypothesized that harmonics can be distinguished only if the frequency separation is greater than the width of a critical band (discussed below). This hypothesis was confirmed by Soderquist (1970) who used in-harmonically related tones to compose his complex sounds.

Parenthetically, it should be pointed out that even experienced listeners have great difficulty in identifying which, of a complex, tones are present, or in determining how many tones are contained in a complex (Thurlow and Rawlings, 1959; Pollack, 1964). It appears that the probe tone in Plomp's procedure aided in selecting components. Such was found to be the case by Thurlow and Bernstein (1957).

Reference was made above to the hypothesis that frequency resolution of the harmonics of complex sounds is limited to the width of the critical band. The notion of the critical band was introduced into auditory theory forty years ago by Fletcher (1940). He assumed that, in detecting tonal signals in noise, the auditory system functions as a bandpass filter the output of which is influenced only by inputs with frequency components that fall within the bandwidth of the filter. This idea was used to account for the differential masking of pure tones by noise (Fletcher, 1940; Hawkins and Stevens, 1950; Bilger and Hirsh, 1956) i.e., as tonal frequency was increased, it was found that the signal-to-noise (S/N) ratio had to be increased in order for the signal to remain detectable at the masked threshold. Fletcher accounted for this by showing that the width of the critical band increased as a function of frequency (see Question 1). Thus, at higher frequencies larger S/N ratios were required to reach the masked threshold because the critical bandwidth was greater and this permitted more noise to mask the signal. Fletcher assumed that, at the masked threshold, the power of the signal equalled the power of the noise contained within the critical band, i.e., $S = WN_0$, where S is signal power, N_0 is spectrum level of the noise, and W is the rectangular equivalent of the critical bandwidth. From this "equal power" assumption, bandwidth was readily calculated from the S/N ratio required for the masked threshold, i.e., $W = S/N_0$ (also referred to as the critical ratio). This was the indirect approach to determinations of critical bandwidth employed by many early investigators. A plot of W (critical ratios) against f shows essentially the same relationship as that found for the absolute frequency difference limen (DL) by Shower and Biddulph (1931) and Harris (1952).

Following Fletcher's example, a number of investigators undertook to measure or estimate the critical bandwidth utilizing a variety of experimental techniques (e.g., Shafer et al., 1950; Webster, et al., 1952; Zwicker, 1952; Gassler, 1954; Zwicker and Feldtkeller, 1955; Bauch, 1956; Hamilton, 1957; Zwicker et al., 1957; Plomp and Bouman, 1959; Scharf, 1959, 1961; Swets et al., 1962; Greenwood, 1961a, b; Jeffress, 1964; Green, 1965; Bourbon et al., 1968; Mulligan et al., 1968; Carterette et al., 1969; Mulligan and Elrod, 1970b; Patterson, 1971, 1974, 1976; Margolis and Small, 1975; and Patterson and Henning, 1977). For the most part, what emerged were functions which, though relatively flat below 500 Hz, increased as a function of frequency. Generally, the obtained relationship between critical bandwidth and center frequency (see Table VI-1) parallel that obtained on the basis of Fletcher's critical ratio, except at frequencies below about 200 Hz. However, the bandwidths obtained by many (e.g., Zwicker et al, 1975) are about twice as wide as the rectangular bandwidths calculated from masked threshold at frequencies

TABLE VI-I. CRITICAL BANDWIDTHS AT THE INDICATED CENTER FREQUENCIES

ΔF	Center and Cutoff Frequencies		ΔF
90	20	65	90
90	110	155	95
95	200	250	95
100	295	345	105
108	395	450	110
120	503	560	130
130	625	690	140
145	755	830	150
160	900	980	175
190	1,060	1,155	200
210	1,250	1,355	225
240	1,460	1,580	255
270	1,700	1,835	295
320	1,970	2,130	350
380	2,290	2,480	420
450	2,670	2,900	500
560	3,120	3,400	620
680	3,680	4,020	760
840	4,360	4,780	920
1,000	5,200	5,700	1,150
1,300	6,200	6,850	1,550
1,800	7,500	8,400	2,100
2,400	9,300	10,500	2,800
3,300	11,700	13,300	4,000
	15,000	17,300	

From Zwicker et al. (1957)

above 200 Hz. The basis for this difference may be that the shape of the auditory filter function is not rectangular (Patterson, 1971, 1974, 1977; Patterson and Henning, 1977), in which case the skirts of the filter function spread over a broader range of frequencies than that encompassed by the equivalent rectangular bandwidth. Patterson's findings indicate that the filter function is symmetrical about its center frequency, dropping off sharply as distance away from the center frequency increases.

Critical bandwidth, however it is obtained, does represent a kind of frequency selectivity that may be manifest in the perception of complex sounds. For example, Plomp and Levelt (1965) showed that tonal consonance can be accounted for in terms of the critical bandwidth. Tones falling within one critical band of each other were found to be judged as dissonant, while tones separated in frequency by more than one bandwidth were judged as consonant. We have already become acquainted with the study by Plomp and Mimpen (1968) who found that, of the 5 to 7 harmonics of complex sounds that may be distinguished by human listeners, the frequency separation of these harmonics must be on the order of one critical bandwidth.

Fletcher (1940) hypothesized that frequency discrimination is related to the critical bandwidth by a constant proportion. If such were the case, it would be expected that bandwidth would depend on the relative intensity of tonal signals in the same way that the frequency DLs of Shower and Biddulph (1931) and Harris (1952) were found to decrease as effective intensity (SL or loudness level) increases. This was the theoretical result from masking data obtained by Mulligan and Elrod (1970b). Rather than a fixed critical bandwidth centered on each frequency, bandwidth varies as a function of both center frequency and S/N ratio. For any center frequency, as S/N ratio increases, auditory bandwidth decreases. It should be noted that Green (1960) obtained evidence suggesting that, for noise signals, the width of the authority passband at any center frequency may increase beyond its minimum (or sum outputs of adjacent bands) to encompass signals that exceed one critical bandwidth.

In summary, it appears that the ear does not obey strictly Ohm's law, as was pointed out by Pollack (1964). Although the auditory system is capable of an extraordinary degree of frequency resolution, as in the case of the DLs obtained by Shower and Biddulph (1931), this capability seems to vanish in the case of complex waveforms such as those employed in some of the experiments reviewed here. It seems that some sort of probe tone is necessary for listeners to identify the 5 to 7 harmonics reported by Plomp and Mimpen (1968). Without such a probe, listeners fail to select out the components of complex sounds (Thurlow and Rawlings, 1959). Furthermore, resolution of harmonics requires that they be separated in frequency by widths in excess of one critical band (Soderquist, 1970). Within critical bands the ear effectively integrates acoustic energy and it seems necessary to take the critical band as the limit on frequency resolution of complex sounds.

7. How is monaural intensity discrimination related to the frequency and intensity of tonal inputs?

Although first examined by Knudsen in 1923, the classical study of monaural intensity difference limens for tones was carried out in 1928 by Riesz. His procedure required listeners to detect the occurrence of beats produced by two tones spaced closely together in frequency, e.g., 1,000 Hz and 1,003 Hz. The intensity of one of the tones would be increased to the point at which the listener noted slow variations in loudness. The magnitude of the intensity increase was taken as the intensity difference limen. Although it is questionable whether the detection of beats is a valid criterion for just discriminable intensity increments, Riesz's findings have continued to be of interest to the present.

Riesz found that the magnitude of the intensity increment required for beat detection varies as a function of the frequency and the intensity above absolute threshold (sensation level in dB) of the primary tone. As the sensation level (SL) increases from about 5 to 40 dB, the size of the relative intensive difference limen (also in decibels) decreases roughly exponentially and then levels off for SL above 40 dB. The rate of decrease depends on the frequency of the primary tone. At any SL, the magnitude of the difference limen is a function of frequency, decreasing in size as frequency increases from 35 Hz to 4 kHz, and then reversing direction for further frequency increments. For example, the relative intensive difference limen for a 35 Hz tone decreases from about 5.5 dB at a SL of 15 dB to about 1.8 dB at a SL of 40 dB, while that for a 4 kHz tone decreases from about 1.4 dB to about 0.5 dB over the same range of SL.

Generally, Riesz's data shows that relative intensity discrimination improves dramatically as intensity increases from low to moderate levels with little further improvement beyond about 40 dB above absolute threshold. However, the magnitude of the relative difference limen at any value of SL depends on frequency, diminishing up to 4 kHz and then increasing with further increments in frequency. At all frequencies above about 70 Hz the relative intensity difference limen is less than 1 dB for intensities greater than about 40 dB above absolute threshold. The general nature of these relationships, although not the precise values, is consistent with the findings of some more recent investigators (Miller, 1947; Tonndorf et al., 1955; Zwicker and Feldtkeller, 1967).

The term "relative difference limen" may leave the reader with some misunderstanding of Riesz's findings if it is not distinguished from "absolute difference limen." In words, the relative difference limen in decibels is ten times the logarithm of the ratio: the intensity increment (ΔI) plus the primary intensity (I) divided by the primary intensity ($10 \text{ Log } (\Delta I + I)/I$). The absolute difference limen in decibels is ten times the logarithm of the ratio of the intensity increment to the absolute threshold ($10 \text{ Log } \Delta I/I_0$). Whereas the relative limen decreases and then levels off as intensity in decibels above absolute threshold increases, the absolute limen increases (at first positively and then nearly linearly) with increases in intensity (Miller, 1947). The linear relationship between the absolute limen and intensity for moderate to high levels indicates that the just detectable intensity increment is a constant ratio of the primary intensity, as required by Weber's law. This relationship also may be regarded as a masking function if ΔI is viewed as the increment in signal power required for detection against a background level I_0. Indeed, the curve on which Miller plotted his intensity DLs for noise stimuli, was originally generated by Hawkins and Stevens (1950) to describe the masking of tones and speech by white noise. The equivalence of masking and intensity (or amplitude) discrimination has been examined more recently in some depth by Henning (1969).

A noteworthy departure from previously reported relationships between the intensity difference limen and intensity of the primary input was reported by Rabinowitz et al. (1976). Based on a summary of the results of a number of studies they observed that intensity discriminability improves as intensity increases up to about 10 dB above absolute threshold, levels off as intensity increases to about 40 dB SL, and then again improves slightly with further increases above 40 dB SL. Similar departures from Weber's law were found for increases in intensity above 25 dB SL by McGill and Goldberg (1968), and by Viemeister (1972).

Perhaps the most interesting departure from Riesz's findings on tone intensity discrimination was reported by Jesteadt et al. (1977). Their procedure required that listeners detect intensity increments in a single tone (pulsed sinusoid) rather than intensity fluctuations (beats) between two closely spaced tones (3 Hz difference) as required in the Riesz study. Jesteadt et al. found that the relationship between the relative intensity difference limen and tonal intensity was independent of

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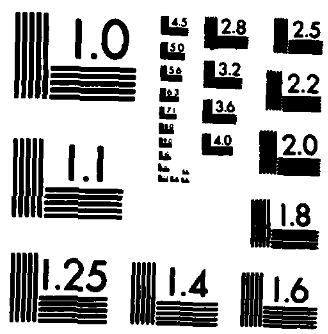

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frequency over the range 200 Hz to 8 kHz. This suggests that the beat detection method employed by Riesz may have introduced a frequency-dependent component into their difference limen data. However, Jesteadt et al. did confirm the general finding of Riesz that relative intensity difference discrimination improves as a function of intensity above absolute threshold, or SL. Even here, however, Jesteadt et al.s' findings differed somewhat from those of Riesz, i.e., the range of their difference limens was smaller and, consequently, the rate of change of the difference limen with increases in SL was not as dramatic. At low values of SL, relative difference limens were smaller than those of Riesz (1.6 dB as compared with 3 dB at 1 kHz and 5 dB SL). At higher values of SL, limens from the two studies were approximately the same (0.5 dB at 80 dB SL). Difference limen values from Jesteadt et al. are given in Table VII-1.

At present it is not entirely clear why Riesz's discrimination data were frequency-dependent while those of Jesteadt et al. were not. In any case, it appears certain that the general relationship between the relative intensity difference limen and sensation level reflects some small improvement in intensity resolution as intensity of input increases. Incidentally, intensity resolution of brief tonal signals has been shown to improve as signal duration increases (Henning, 1970).

Based on the values given in Table VII-1, it can be expected that intensities differing as much as about 1 dB may be monaurally resolved if the input intensity is greater than about 40 dB SL under quiet listening conditions. It should be noted here that Hershkowitz and Durlach (1969a)

TABLE VII-I.

Tabled values determined from equations used by Jesteadt et al. (1977) to fit their data: $10 \log [(\Delta I + I)/I] = 1.644 - 0.0141 \times 10 \log (I/I_0)$; and $\Delta I/I = 0.463(I/I_0)^{-0.072}$. $SL = 10 \log (I/I_0)$.

SL	$10 \log [(\Delta I + I)/I]$	$\Delta I/I$
5	1.57	0.44
10	1.50	0.41
15	1.43	0.39
20	1.36	0.37
25	1.29	0.35
30	1.22	0.32
35	1.15	0.30
40	1.08	0.28
45	1.01	0.26
50	0.94	0.24
55	0.87	0.22
60	0.80	0.20
65	0.73	0.18
70	0.66	0.16
75	0.59	0.14
80	0.52	0.13

reported minimally discriminable binaural intensity difference limens on the order of 0.8 dB (see Question 4). Discrimination of binaural intensity differences is, thus, approximately of the same degree as monaurally discriminable intensities. It is interesting to note that, in the case where no interaural intensity difference exists, discrimination of successive intensity changes at both ears is better (i.e., the signal-to-noise ratio can be reduced without lowering performance) if the signals are presented out-of-phase at the two ears (Henning, 1973).

The foregoing discussion has dealt primarily with the question, given a tone at some frequency and sensation level, by what amount must its intensity be increased in order for a listener to detect the increment? One might also ask, by what amount must two tones of different frequency differ in intensity in order for a listener to discriminate among them on the basis of loudness? This question has not been addressed directly. However, a reasonable guess can be made for tone of frequencies between about 20 Hz and 15 kHz by consulting the equal-loudness contours of Fletcher and Munson (1933) or those of Robinson and Dadson (1956). The former apply to earphone listening while the latter apply to free field listening. Since these contours indicate the sound pressure levels (SPLs) necessary to achieve a loudness level match between tones of different frequencies, they may also be used to ascertain the SPLs of tones that will not result in a loudness match, i.e., whether the tones will be of different loudness levels. For example, Robinson and Dadson's contours indicate that 1 and 2 kHz tones differ in loudness level by 2 dB if the SPL of each is 30 dB. Since the relative DLs of these two tones calculate to be 1.28 dB and 1.25 dB respectively, it may be expected that the loudness difference of 2 dB would be discriminable. This technique should be used only for obtaining rough indications of the SPLs needed to achieve different loudness levels (not loudness difference limens) for tones of different frequencies. The relationship of intensity DLs to loudness has not been established, although Durlach and Braida (1969) have undertaken a theoretical analysis of the whole problem of intensity perception.

8. What cues are available for monaural discrimination among the loudnesses of complex sounds?

If we regard a pure tone, which may be described as a sinusoidal function of time, as a simple sound, then by comparison a waveform composed of even two trignometrically summed sinusoidal functions different in frequency must be regarded as complex. The sounds of bells, buzzers, musical instruments, machines, human voice, etc., are all acoustically complex. In order to achieve some degree of generality of results and to facilitate manipulation of experimental variables, researchers interested in the study of loudness of complex sounds typically have restricted their stimuli to compositions of tones or continuous bands of noise. It has been found that loudness varies not only as a function of intensity, but also spectral and temporal variables.

Beyond such findings as those reviewed in Question 7, which apply to tones and white noise, little can be said regarding discriminability in the intensive domain of sound. Presumably, the same underlying processes would operate with complex sounds. However, the relatively straightforward procedure for determining an intensity difference limen becomes complicated if the acoustic inputs to the listener differ physically in ways that permit discrimination on the basis of cues other than intensity. For example, if two acoustic spectra differ in harmonic structure, a listener may readily discriminate between the two on the basis of timber as is done in the case of sounds produced by different musical instruments (Risset and Mathews, 1969; Grey and Moorer, 1977). Even this appears to be an oversimplification since the spectral characteristics of the sounds of musical instruments vary over time and require of the listener some sort of abstraction to separate temporal and structural characteristics (Huggins, 1952; Miller and Carterette, 1975). Consequently, sounds of complex spectra and temporal patterns generally have not been used in studies aimed at evaluating intensity discriminability. Rather, the direction taken has been aimed at evaluating such sounds in terms of loudness.

In the case of sounds consisting of either multi-component or continuous spectra, loudness depends on the manner in which the auditory system sums the component intensities. The phenomenon of monaural loudness summation in the spectral domain is based in the finding that the loudness of a broad spectrum sound (e.g., noise) increases as its bandwidth increases beyond a certain point, even though the overall sound pressure level remains constant (Zwicker and Feldtkeller, 1955). The point at which loudness summation begins is the critical bandwidth (Zwicker et al., 1957; see Table VI-1).

Obviously, if the bandwidth of a noise is increased, the only way its overall level can be held constant is by reducing its average level per cycle, i.e., its spectrum level. This is why loudness of the band of noise does not increase so long as it is no wider than the critical band in which it is centered, i.e., since spectrum level is reduced in direct proportion to increments in noise width, the integral of the acoustic power within the critical band remains constant so long as the input bandwidth does not exceed the critical bandwidth. The critical band is a frequency region over which acoustic power is summed (see Question 6) without regard for spectral shape or width. However, summation of acoustic power within a critical band and loudness summation outside critical bands are different processes.

In Zwicker and Feldtkeller's experiment, if spectrum level had been held constant while increasing noise bandwidth, then the overall power within the critical band (and loudness) would have increased in proportion with increases in the noise bandwidth until the latter equalled the width of the critical band. Even with further increases in noise bandwidth beyond this point, no further increase in power would have occurred within the critical band so long as the noise spectrum level had remained constant. However, loudness would have continued to increase as the noise bandwidth exceeded the width of the critical band. It is this phenomenon to which the notion of loudness summation applies, not the increase in loudness which occurs as a result of power summation within a critical band. The latter was eliminated in Zwicker and Feldtkeller's experiment by keeping overall level of the noise constant, i.e., increases in bandwidth were accompanied by decreases in spectrum level.

Zwicker and Feldtkeller found that loudness increases as a function of the frequency width (greater than that of a critical band) of complex sounds even when the overall power is held constant. This finding seems the more remarkable considering that, as pointed out above, in order to keep the overall power constant, the power at each frequency throughout the noise band (spectrum level) had to be decreased in direct proportion to increases in its frequency width. Thus, as the noise bandwidth was increased and its spectrum level was decreased, loudness increased! Similar findings have been reported by Scharf (1970a) for sounds consisting of but two frequency components.

Scharf (1962) has shown also that the shape of spectra may influence loudness summation. For example, if each component of a multiple complex is of the same loudness level, the loudness of the complex will be greater than in the case of a spectrum with the same overall intensity but having components of unequal loudnesses. Similarly, Zwicker et al. (1957) found that the frequency intervals between multitone components was a determiner of loudness. Tones spaced at intervals of one critical band were found to be louder.

The foregoing suggests that a wide band sound would be judged to be louder than a sound with a more narrow bandwidth if the overall power in the two bands were equal. Conversely, in order for the two sounds to be judged equally loud, the overall level of the sound with the more narrow bandwidth would have to exceed that of the wider band sound. Furthermore, the frequency spacing and relative intensities of the components of wide-band complex sounds would be expected to influence overall loudness, since insofar as loudness is concerned, the auditory system ignores the spectral distribution of acoustic energy <u>only</u> within critical bands.

It is interesting to note that loudness summation does not appear to occur if overall levels are no more than 10 to 15 dB above absolute threshold (Scharf, 1959). This is probably due to the fact that, at such overall levels, the spectrum levels of wide band sounds would be near threshold. It also seems to be the case that loudness summation is greatest at moderate levels (Pollack, 1952) between 40 and 60 dB. Zwicker and Feldtkeller's data indicate that, at overall levels above 60 dB, the rate of increase in loudness is somewhat less than at lower levels. It is, perhaps, more than an interesting coincidence that the loudness function for white noise differs from that of the 1 kHz tone in much the same way. Since the threshold SPL for noise is somewhat greater than the 1 kHz tone threshold, the noise loudness function grows more rapidly than the tone function, crossing it in the vacinity of 22 dB SPL. At this point loudness of the tone and noise are the same, even though the bandwidth of the noise exceeds one critical band. As SPL increases beyond this point, loudness of the noise continues to grow more rapidly than that of the tone up to about 60 dB SPL. Above this level, the rate of growth of noise loudness declines and the two functions converge. Loudness values for these two functions are given in Table VIII-1. These relationships have been obtained both by the method of magnitude estimation and production (Scharf and Fishken, 1970) and by loudness matching (Hellman, 1976). It appears that the rate of growth of noise loudness varies in much the same way with increases in SPL as it does with increases in spectral width beyond one critical band.

TABLE VIII-1. SONE VALUES FOR 1000-Hz TONE AND FOR WHITE NOISE
(Scharf and Fishken, 1970)

SPL dB	Tone	Noise	SPL dB	Tone	Noise	SPL dB	Tone	Noise
10	.052	-	50	2.00	3.85	90	32.0	46.0
12	.072	-	52	2.30	4.45	92	36.8	50.5
14	.095	-	54	2.64	5.20	94	42.2	57.5
15	.110	-	55	2.83	5.60	95	45.3	61.0
16	.125	-	56	3.03	6.00	96	48.5	65.0
18	.155	-	58	3.48	7.00	98	55.7	72.0
20	.190	-	60	4.00	7.85	100	64.0	80.0
22	.230	-	62	4.59	8.90	102	73.5	91.0
24	.280	-	64	5.28	10.20	104	84.4	102.0
25	.305	-	65	5.66	10.90	105	90.5	108.0
26	.330	-	66	6.06	11.50	106	97.0	114.0
28	.395	.450	68	6.96	13.00	108	111.0	128.0
30	.460	.580	70	8.00	14.70	110	128.0	-
32	.550	.720	72	9.19	16.40	112	147.0	-
34	.640	.900	74	10.60	18.50	114	169.0	-
35	.700	1.000	75	11.30	19.50	115	181.0	-
36	.750	1.100	76	12.10	20.60	116	194.0	-
38	.860	1.360	78	13.90	23.20	118	223.0	-
40	1.000	1.650	80	16.00	26.00	120	256.0	-
42	1.115	2.000	82	18.40	29.00			
44	1.320	2.400	84	21.10	32.50			
45	1.410	2.600	85	22.60	34.80			
46	1.520	2.800	86	24.30	36.50			
48	1.740	3.280	88	27.90	41.00			

(From Scharf (1978, p. 196)

Studies of the influence of temporal variables on loudness have focused mainly on effects of the duration of single sounds and the interval separating successive pairs (or trains) of brief sounds. The abruptness of sound onset (rise time) also has been shown to influence the loudness of wide band noise and tones (Gjaevenes and Rimstad, 1972), provided rise time is brief (e.g., 30 msec) and the sound contains low frequency components (e.g., 250 Hz).

Generally, as the duration of exposure to a sound increases from a few milliseconds, its loudness rapidly increases to some steady level within the first 50 to 500 msec. It is unclear what factors determine the critical duration beyond which no further growth in loudness occurs. Scharf (1978) provides a summary of findings from a number of studies in which tones and noise were used, but there appears to be little consensus on such matters of interest as the trading relation between intensity and duration, the dependence of the critical duration on intensity, frequency, bandwidth, etc. As shown by Port (1963), growth in loudness within the first 50 msec of continuous exposure may be considerable. At the other extreme where decreases in loudness with long durations of exposure would be found, it appears that if any loudness adaptation occurs at all, it is limited to exposure levels near threshold even for exposures as long as 30 min. The reader is referred to Scharf (1978) for a review of the pertinent literature. As Scharf points out, adaptation to the loudness of intense sounds " . . . would be comforting but costly" in the preservation of normal hearing.

As the interval separating a pair of tone pulses of the same frequency (10 msec duration; 5 msec rise-decay) increases from several milliseconds to some upper limit, the combined loudness of the pair as well as the loudness of the second pulse in the pair, decrease (Irwin and Zwislocki, 1971). The upper limit of the inter-pulse interval beyond which enhancement of loudness of the pair no longer exceeds the loudnesses of the indivudal pulses presented alone has been found to lie between about 25 and 200 msec (see Scharf, 1978, for a review of this literature).

Consistent with the findings on spectral contributions to loudness summation, Scharf (1970b) found that the combined loudness of pairs of tone pulses (5 msec duration; 1.5 msec rise-decay) separated by time intervals of up to about 200 msec increased as the frequency difference between individual pulses increased. As Irwin and Zwislocki (1971) demonstrated, the increase in loudness of pairs of pulses of different frequency exceeds that for pulse pairs of the same frequency (or the second number of such pairs). The former effect is thus considered to be summation rather than enhancement.

In the case of trains of successive pulses of duration T, as the pulse rate decreases from its maximum at I/T to about 2 pps, the level of the pulsed sound required to maintain constant loudness decreases (Pollack, 1958). Presumably, an interrupted sound would be judged louder than a continuous sound of the same spectrum and overall level.

9. Are reaction times to acoustic stimuli dependent on loudness?

In an early study by Chocholle (1940) reaction times to tonal stimuli were found to be a function of both intensity and frequency. However, Chocholle showed that the frequency dependency could be eliminated by equating the loudness of tones of different frequency. Once this correction for differential sensitivity was made, the data clearly showed that reaction times to sounds of equal loudness (not intensity) are equal even though the sounds may differ greatly in frequency. This fact enabled Chocholle (1954) to collapse data obtained at frequencies ranging from 50 Hz to 10 kHz on to a single function relating reaction time to loudness level in phones. The function shows that, for any frequency, as the loudness of the acoustic stimulus increases, the latency of the reaction to it (RT) decreases roughly exponentially. The magnitude of the reduction in RT is from approximately 300 msec at about two phones to approximately 110 msec at 90 phones, nearly a 3-fold increment in speed of reaction that approaches what appears to be the upper limit on the capacity of humans to respond. Incidentally, speed of reaction to binaural signals is faster than for monaural signals (Chocholle, 1946). Also, the relationship between reaction time and loudness is more steep (RT changes faster) in the presence of masking noise than in quiet (Chocholle and Da Costa, 1971; Chocholle and Greenbaum, 1966; also, see the discussion of Kohfeld's, 1971, study in question 19).

10. What are the joint effects of time uncertainty, S-R compatibility, and stimulus intensity on auditory choice-reaction times?

In a study by Sanders (1977) involving a 3-signal 3-response arrangement, it was found that the joint effects of time uncertainty, S-R compatibility, and stimulus intensity on choice-reaction times to auditory signals are additive (see question 18). No significant interaction effects were obtained between any combination of these variables. The fastest RTs were obtained under the following combination of conditions; no time uncertainty of signal presentation, high S-R compatibility, and high stimulus intensity. The slowest RTs were obtained with maximal time uncertainty, low compatibility, and low stimulus intensity. Increases in stimulus intensity (from 35 dB to 85 dB) resulted in less reduction of RTs than increments in signal certainty and S-R compatibility. The results of this study suggest that time uncertainty and S-R compatibility can be traded against each other, e.g., temporal uncertainty of a signal may be compensated for by making it highly compatible with the response, especially in a choice-reaction situation. Further improvement can be obtained by increasing signal intensity. Apparently, the latter effect is peculiar to the auditory modality since increases in intensity of visual signals has not been found to offset time uncertainty (Sanders and Wertheim, 1973; Bernstein et al., 1973).

11. How may stimulus-response compatibility be adjusted for optimum reaction times?

In any situation requiring an optimally rapid response to some particular stimulus occurrence, the stimulus must convey to the operator certain minimal information. For this reason, we prefer to speak of signals rather than stimuli.

If any one of a number of responses to onset of a signal is acceptable - whether equally effective or not - the signal information is simply "react now." In this case the signal is merely a time marker. If just one response is acceptable, the signal information is "make the appropriate response now." Here the signal must not only mark the start time of the appropriate response, it must select the response as well. This selective function becomes more informationally loaded if the operator must make a choice, i.e., if more than one signal-response association is required. For example, if two tones are each associated with a different response, the occurrence of either signal demands an immediate choice, even if it is an automatic one in a well-practiced operator. The information processing approach to the study of choice-reaction times has provided a theoretical basis for understanding the relationship between RT and the number of alternative choices available to an operator (Hick, 1952; Hyman, 1953; Smith, 1977). It is in such choice situation that the problem of signal-response compatibility is most significant (Smith, 1977), although it is present even in the one-signal, one-response situation.

The notion of stimulus-response compatibility is vague, if not misleading, even though it has acquired the status of an experimental variable. Broadbent (1971), for example, would define compatibility in terms of the "obviousness" of the correspondence of stimuli and responses. As Duncan (1977) has pointed out, however, "obvious" correspondence may exist between stimuli and responses for a given individual due to extensive practice. From this point of view, compatibility could not be defined completely in terms of objective dimensions, e.g., spatial relations.

One may be tempted to define S-R compatibility operationally in terms of average reaction time, e.g., stimuli and responses are compatible if, when paired, they yield average RTs that are smaller than other S-R combinations. This is hardly satisfactory. Not only would this definition not distinguish among compatibility and other variables that may be manipulated to obtain small RTs, but it would not allow application of the notion of compatibility to other response measures.

It appears that no generally applicable and explicit definition of S-R compatibility has been adopted in the literature. The approach that is taken in most studies is simply one of naming certain experimental conditions as "compatible" and "incompatible," where the difference apparently is presumed to be obvious. In some cases it is. For example, in a study by Broadbent and Gregory (1965), tactual stimuli were presented to the operator's fingers. In the high compatibility condition, the finger stimulated was the one to respond. In the low compatibility condition, the finger stimulated corresponded with the one to respond located on the other hand. Here, S-R compatibility appears to mean proximity of the site of stimulation to the responding organ.

A similar sort of S-R compatibility was employed by Sanders (1977). Three lamps were placed in a row with a response key located immediately beneath each one. The operator's preferred hand was placed so that the three middle fingers were positioned over the response keys - one finger assigned to each key. In the compatibility condition, the appropriate response was signalled by illumination of the lamp located directly above the correct finger-key position. In the incompatible condition, the spatial relationship between lamp and finger-key position did not correspond; the left and middle lamps signalled responses of the middle and right fingers respectively, while the right lamp signalled a response of the left finger. Again, as in the study by Broadbent and Gregory (1965), S-R compatibility seems to mean proximity. However, this is not proximity of the site of stimulation to the response organ since the signal is received visually. Rather, this is a kind of relative proximity derived from the relative locations of response organs and the relative points of origin in visual space of the signals.

A comparable form of spatial S-R compatibility was extended to an arrangement of auditory signals by Sanders (1977). In this case the signals were bursts of noise delivered either to the left ear alone, to both ears, or to the right ear alone. Compatibility was defined as in the visual signal experiment above. The left, middle, and right finger responses were indicated by noise signals to the left ear, both ears, and right ear respectively. Incompatibility was established just as it was in the previous experiment, except with auditory signals. Of course, the signals presented singly to each ear were localized at one side or the other, while presentation of the signals to both ears simultaneously (interaural correlation + 1) were localized medially. This resulted in a distribution of signals in auditory space that corresponded to the left, middle, and right finger positions. In this case, S-R compatibility does not involve visually perceived spatial proximity but rather the perceived correspondence of spatial orders. It is of interest that S-R compatibility in the latter case resulted in an improvement in RTs that was approximatey of the same magnitudes as that obtained in the visual proximity experiment.

To return to the original point, it is the information conveyed to the operator by the signal that appears to determine the speed of choice reaction. In choice situations, the notion of S-R compatibility becomes especially important for achieving fast reaction times because, in addition to marking the time-to-respond, the signal must select the appropriate response. If S-R compatibility is high, then the signal provides definitive information identifying the response to be made. This requires some clear relations between responses and signals, usually spatial. Evidently, it is the spatial relationship which is most obviously (in Broadbent's sense) either compatible or not.

Duncan (1977) has shown that there are two properties of S-R spatial relationships ("mapping") that are important for RTs, viz., the relationship that exists for each S-R pair, and the set of such relationships for all S-R pairs in a situation. RTs are fast if the same spatial relationship holds for all S-R pairs in a set.

A somewhat different sort of compatibility has been designated as "ideomotor" (IM) by Greenwald (1972). In IM, the compatibility relationship is assumed to exist between the stimulus and the sensory feedback from the response. If the feedback "resembles" the stimulus, then compatibility is high. Some examples from Greenwald and Shulman (1973) are: the movement of a switch (right or left) in the direction indicated by a visually displayed "arrow"; vocal reproduction of the sound of a letter (e.g., "A" or "B") presented through headphones.

The example of the response to the "arrow" appears to be an instance of what Audley et al. (1975) referred to as a "symbolic S-R code," e.g., a response to the left if the number "1" is presented and a response to the right if the number "2" is presented. In such "symbolic" cases, the assumption that response feedback "resembles" the stimulus would seem to be untenable. However, such an assumption may be valid in the case of vocal matching of speech signals. It may be valid also in such cases as matching the sound of one instrument to that of another, matching visual patterns, etc. The response might produce the matching stimulus directly (e.g., vocal), or indirectly through operation of a device (e.g., sounding a note on a musical instrument or alignment of two visual patterns). In such cases, it is the sensory consequences of the response that match the sensory effects of the signal. It would thus seem that this form of compatibility would be better termed SS than IM. In the case of symbolic S-R compatibility, it is difficult to understand in what manner response feedback could be said to match the stimulus.

In all forms of S-R compatibility, the signal specifies the response. It may select the response organ directly, identify the response spatially, provide a pattern to be matched by the response or its environmental consequence, or it symbolically calls forth the response.

12. Is speed of reaction to stimuli influenced by the probability of stimulus occurrence?

Generally, the higher the probability of stimulus occurrence, the faster the reaction to it will be. According to Audley et al. (1975), high stimulus probabilities produced "anticipations" which result in short average reaction times. It should also be noted that Audley et al. (1975) found that error responses are faster (shorter RTs) than correct responses, a result that is consistent with those of others, e.g., Yellott (1971). If, as in experiments by Audley et al., two stimuli are presented and one is associated (through an advance numerical cue to the subject) with a high probabiliy, the average RT to that stimulus will be shortened while the RT to the uncued stimulus will either be unaffected or increased. These findings are consistent with those of Sanders (1977), as well as others, who have investigated the effects of time uncertainty on reaction time. The less time uncertainty, i.e., the greater the probability that the stimulus will be presented at a point in time following a warning signal, the faster the average RT. In a series of studies concerned with the influence of expectancy on simple RT, Naatanen and Merisalo (1977) found "subjective probability" followed the objective probability of stimulus occurrence. They concluded that expectancy, or "subjective probability" was ."..the most important determinant of preparation and, hence, of the reaction time...."

13. Is speed of vocal identification faster for geometric symbols than for numerals?

In a study by Forrin and Morin (1967), it was found that the average time required to name (vocal response) visually presented numerals was increased if geometric symbols assigned arbitrary names were introduced into the numerical stimulus sequence. This finding was further explored by Forrin (1975) in a subsequent study utilizing much the same methodology. The overall outcome confirmed the earlier result, i.e., naming of numerals in sequence was faster than if the sequence contained symbols. However, symbol-symbol sequences evoked faster correct reaction than did symbol-numeral sequences. The shortest latencies were obtained for sequences in which the same numeral was either repeated or all stimuli were numerals (numbers of the same class). The presentation of a warning signal prior to the stimulus resulted in a reduction of latencies to all stimulus sequences.

Since the stimuli used in these experiments were visual, it is unclear whether the same relations can be expected for equivalent auditory stimuli. However, it would seem not far-fetched to expect that if RTs to one class of auditory stimuli (complex) are slower than RTs to another (simple), then alternative presentation of the two would probably result in an increased RT for the simple stimulus class.

14. Are reactions to stimuli of a symbolic nature (triangles, arrows, etc.) influenced by the order of presentation of such stimuli?

It appears that both the mean reaction time and the error rate of responses to symbolic stimuli are affected by the order in which such stimuli are presented. For example, Falmagne et al. (1975) found that the reaction (in terms of latencies and errors) to one of two stimuli was conditional upon the sequence of these two stimuli that preceded presentation of the one to which the reaction was made. That is, the reaction to stimulus "1" depended on whether it was preceded by presentations of "1" or "2." Falmagne et al. (1975) reported that the RT and error rate of the response to a particular stimulus was faster and larger, respectively, if the immediately preceding stimulus were the same, i.e., if the stimulus to be reacted to were a repetition. If the stimulus were not the same as the one that preceded it, the average RT would be slower and the error rate would be lower. Stimulus probability was also found to exert an influence, e.g., reaction times to more probable stimuli were faster and error rates generally greater. Although these effects do not appear to have been investigated using auditory stimuli, there is every reason to think that the same sort of sequential dependencies would hold.

15. Do choice-reaction times increase as the number of signals (choices) increase?

Early models of choice-reaction time data (Hick, 1952; Hyman, 1953) assumed that RT increased as a linear function of the logarithm of the number of stimulus-response choices. Hence, RTs in a 8-stimulus, 8-response situation would be slower (longer latencies) than in a 2-stimulus, 2-response situation. This no longer appears to be a complete account. Leonard (1959) found that, with a high degree of stimulus-response compatibility, RTs did not increase as the number of choices increased beyond 2 for practiced operators. In this study a high level of S-R compatibility was achieved by delivering the signal (a vibro-tactile stimulus) directly to the finger that was to respond. In a similar, but more carefully controlled experiment, Smith (1977) confirmed Leonard's findings and further showed that RTs tend to increase as a function of number of choices only under conditions involving low S-R compatibility, or mixtures of low and high compatibility choice-reactions. Smith also employed vibro-tactile stimulation as his signals. At present, it is not clear whether a sufficiently high S-R compatibility can be achieved for auditory signals greater in number than 3 (as in the study by Sanders, 1977) to overcome the slowing effect imposed by an increase in the number of choice-reactions. If not, RTs may be expected to increase as the number of auditory choice-reactions increase.

16. If an operator is prepared for rapid reaction to one stimulus, will his reaction time to a second stimulus be affected?

Although the answer to this question cannot be given concretely on the basis of presently available data, studies by Audley et al. (1975) suggest that there is an effect. Preparation to respond to one of two stimuli was found to result in a decrease in the RT for that stimulus-response. However, the RT for the other S-R increased. A plot of mean RTs to each stimulus against each other, i.e., the means to one stimulus plotted on the ordinate and those for the other plotted on the abscissa, yields a function which Audley calls a "time exchange relation." Most of the exchange relations obtained by Audley et al. (1975) were linear. In his words, this indicates that ."..the decrease in reaction time due to being prepared for one stimulus or its response, must equal the increase in reaction time to the other stimulus." Based on his analysis of data from Remington (1969) and Schvaneveldt and Chase (1969), Audley et al. hypothesized that the shape of the exchange relation depends on the stimulus-response code (symbolic S-R compatibility). They utilized two such codes (one spatial and one numerical) but did not find any dependence of the exchange relation on code type.

17. What are the primary determinants of reaction times of successive responses to repeated events?

In the case of repetitions of the same response to the same stimulus, a major variable shown to influence reaction times is the length of the time interval between the end of the response and the onset of the next stimulus in the series, i.e., the R-S interval (Smith et al., 1973). However, studies limited to repetitions of the same stimulus-response pair do not provide a basis for generalization to more "real life" situations that involve different reactions to various stimuli. As pointed out by Rabbitt et al. (1975), in order to obtain the data necessary to determine how successive responses are programmed, experimental designs should require that subjects choose between responses that differ in "nature"; the signals should be equally discriminable and paired with responses in all possible S-R mappings. The number of such stimuli should also be fixed to keep the informational load constant. These controls should permit valid comparisons for the assessment of response programming in choice-reaction time experiments.

This approach was taken in a study by Rabbitt (1966) in which subjects responded to four different signals with responses of both hands and feet. As in other studies, repeated responses were found to be fastest. In mixed sequences of responses, it was found that both the limb type (foot or hand) and the laterality of limbs were important. For example, if a response by one limb were followed by a response of a limb of the same type on the opposite side of the body (right foot response then left foot response; or response of the right hand followed by a response of the left hand), faster reactions were made than if the contralateral limb were of a different type (right foot response then left hand response, etc.). The worst case occurred for successive responses by different limbs on the same side of the body (right foot response followed by right hand response, etc.). This condition resulted in the largest RTs and error rates. Evidently, the nature of the transitions that must be made from response to response in successive choice-reaction situations is a major determiner of both speed and accuracy.

A slightly different picture is obtained if responses are limited to the hands alone. In a study by Rabbitt and Vyas (1973), finger responses on keyboard tasks were examined under conditions involving choices between the hands. It was found that the faster responses were made when the succession of fingers were on the same hand. The response of a finger on one hand was always slower if it was preceded by a finger response on the other hand. Unlike transitions from foot to hand on the same side of the body, ipsilateral transitions among the fingers were found to be facilitory. However, even faster RTs were obtained for successive responses of corresponding fingers on the two hands (left index finger follows right index finger, etc.). Rabbitt and Vyas concluded that ."..successive, different responses are made more quickly if they involve successive use of the same hand or of the same finger on different hands."

In a more recent series of studies, Rabbitt et al. (1975) attempted to evaluate the influence of response complexity on reaction time, where complexity was defined in terms of the number of fingers and hands required to make a response. The main finding was that the more complex a response on one trial, the slower the RT of the response on the following trial. They also found that the time taken to respond with any one finger depends on the number of other fingers that must be coordinated with it. Furthermore, transitions from response to response in succession were found to be more difficult if different hands were required. It appears that the evaluation of response complexity must take into account the organization of the movements comprising a response as well as the nature of the changes in organization that must occur during a transition from one response to the next in a succession of choice responses.

These investigators also found evidence for another ramification of response complexity. Subjects were required to validate their responses, i.e., to indicate whether a response was thought to be correct or incorrect. The more complex responses required longer times to validate, and incorrect responses were validated more quickly than correct responses.

18. What task variables have been found to add or interact in their effects on reaction times?

Stimulus-response compatibility has been reported to be additive in its effect on reaction times (mean RTs) with the following variables: signal degradation (Sternberg, 1969); signal discriminability (Rabbitt, 1967); time uncertainty of signal presentation (Sanders, 1977); stimulus intensity (Sanders, 1977); motor preparation (Sanders, 1970); and fore-period duration (Posner et al., 1973). However, the joint effects of S-R compatibility and either signal degradation or discriminability may disappear with practice (Sanders, 1977). S-R compatibility has been found to interact in its effect on RTs with the time uncertainty of signal presentation (Broadbent and Gregory, 1965; Sternberg, 1969), the number of signal alternatives (Brainard et al., 1962), the relative frequency of signal occurrence (Fitts et al., 1963; Sanders, 1970), and practice (Sanders, 1977).

The degree to which signals are degraded has been shown to be additive in its effect on RTs with the following variables: S-R compatibility (cited above); the relative frequency of signal presentation (Miller and Pachella, 1973); and the time uncertainty of signal presentation (Posner et al., 1973). The effect of signal degradation on RTs also has been found to interact with the number of signal alternatives (Sternberg, 1969) and the relative frequency of signal presentation (Miller and Pachella, 1973). The latter variable, relative frequency, interacts with two others as well, viz., time uncertainty (Bertelson and Barzeele, 1965) and motor preparation (Sanders, 1970).

A variable related to both signal discriminability and degradation is signal intensity. The latter has been found to be additive in its effect on RTs with both time uncertainty (Raab et al., 1961) and sensory modality (Howell and Donaldson, 1962). The additivity of the effect of time uncertainty with that of other variables has been reported not only for stimulus intensity (cited above) but also for the number of signal alternatives (Alegria and Bertelson, 1970; Broadbent and Gregory, 1965), signal degradation (cited above), and signal detection (Egan et al., 1961).

19. Are reactions to auditory stimuli faster than reactions to visual stimuli?

Although modality comparisons of reaction times have been reported in scattered studies, it appears that little systematic attention has been devoted to this matter. One prominent difficulty involved in such a comparison is the determination of functionally equivalent categories of auditory and visual stimuli. Unless these two energy forms are equated on some dimension that is relevant to the task, differences in reaction times to the two types of stimuli are difficult, if not impossible, to evaluate. The general finding, therefore, that RTs to auditory stimuli are faster than RTs to visual stimuli is in need of further systematic study, and traditionally the explanation for this (i.e., auditory stimuli are more arousing) must be regarded as merely suggestive. Oddly enough, one of the best studies to date on the question of auditory _vs_. visual RTs was designed to examine the effects of drugs on speed of reaction (Trumbo and Gaillard, 1975). As a matter of design, Trumbo and Gaillard made their auditory and visual stimuli informationally equivalent. They found that, under all conditions of time uncertainty and drug treatments, auditory RTs were faster than visual RTs on the order of 40 to 50 msec, although both types varied in essentially the same manner as a function of the experimental parameters. A modality-drug interaction was obtained which lends some credence to the arousal hypothesis mentioned above. The barbituric treatment resulted in an increase in auditory RT while having negligible effect on visual RT. This would be the expected result if the major difference in auditory and visual RTs can be attributed to differences in the arousing potential of the two types of stimuli. By contrast, amphetamine affected only visual RTs, i.e., shortened them.

In a study designed to determine the relationship between stimulus intensity and RT, Kohfeld (1971) equated the decibel value of light and sound stimuli. The range of stimulus values varied between 30 and 90 dB. Reference values for light and sound stimuli were 10^{-10} lambert and 0.0002 dines/cm^2 respectively. Whether or not these particular reference values effectively equated decibel values of the light and sound stimuli, Kohfeld found that, as intensity in dB increased RTs to both kinds of stimuli decreased. However, RTs to the light stimuli decreased more rapidly than did RTs to the sound stimuli and, consequently, the two functions converged at high intensity levels. At low to moderate intensity levels, RTs to auditory stimuli were approximately 20 to 30 msec faster than RTs to visual stimuli of the same decibel value. This difference disappeared in the vicinity of 50 dB, the cross-over point from scotopic to photopic vision. Hence, intense light stimuli in the photopic range of vision appear to be as effective as sounds in initiating fast reactions. As an afterthought, it would seem that a better way for equating the intensive diminsion of light and sound stimuli would be by means of magnitude estimation in which a cross modality procedure were used to match brightness and loudness of the respective stimuli. Even if the intensive dimensions are matched, however, the rise times of sound stimuli may play a far more pertinent role in affecting RTs than the rise times of light stimuli.

APPENDIX I.

LIST OF QUESTIONS WITH REFERENCES

1. What factors determine the detectability of monaural signals?

(Fletcher, 1940; Hawkins and Stevens, 1950; Shaw and Piercy, 1962; Licklider, 1951; Davis and Krantz, 1964; Davis et al., 1950; Stevens and Davis, 1938; Egan and Hake, 1950; Bilger and Hirsh, 1956; Scharf, 1959, 1961, 1971; Garner, 1947a, b; Garner and Miller, 1947; Zwislocki, 1960; Zwislocki and Pirodda, 1952; Luscher and Zwislocki, 1949; Corso, 1958, 1963, 1967; Ward, 1963, 1966, 1970; de Mare, 1939; Olsen and Carhart, 1966; Mulligan et al., 1967; Mulligan and Adams, 1968; Mulligan et al., 1968; Mulligan and Elrod, 1970; Mills et al., 1970; Mosko et al., 1970; Munson and Gardner, 1950; Riach et al., 1962; Hickling, 1967; Elliot, 1962a, b; Gardner, 1947; Stein, 1960; Raab, 1961; Osman and Raab, 1963; Robinson and Pollack, 1971; Wright, 1964; Shafer et al., 1950; Webster et al., 1952; Zwicker et al, 1957; Greenwood, 1961; Green et al., 1959; Swets et al., 1962; Bourbon et al., 1968; Carterette et al., 1969; Patterson, 1971, 1974, 1976; Patterson and Henning, 1977; Margolis and Small, 1975; Blodgett et al., 1958; Hirsh and Burgeat, 1958; Durlach, 1972)

2. What factors determine the detectability of binaural signals?

(Licklider, 1948; Hirsh, 1948; Hirsh and Webster, 1949; Hirsh and Burgeat, 1958; Webster, 1951; Jeffress et al., 1952, 1962; Jeffress et al., 1956; Blodgett et al., 1958; Blodgett et al., 1962; Durlach, 1963, 1972; Colburn and Durlach, 1965; Rilling and Jeffress, 1965; Robinson and Jeffress, 1963; Goodnow and Jeffress, 1957; Bourbon and Jeffress, 1965; Egan, 1965; McFadden, 1968; Langford and Jeffress, 1964; Rabiner et al., 1966; Mulligan (in press); Mulligan et al., 1967; Mulligan and Wilbanks, 1965; Mulligan and Cornelius, 1972; Dolan and Robinson, 1967; Wilbanks and Whitmore, 1968; Whitmore and Wilbanks, 1965a, b; Mills, 1960, 1972; Schenkel, 1964; Schubert, 1956; Sondhi and Guttman, 1966; Metz et al., 1968; Wightman, 1971; Wightman and Houtgast, 1972; Weston and Miller, 1965)

3. On what does the localization of sound sources in "auditory space" depend, and what are its limits?

(Woodworth and Schlosberg, 1954; Feddersen et al., 1957; Shaw, 1974a, b; Blauert, 1974; Hershkowitz and Durlach, 1969a; Domnitz, 1973; Mills, 1958, 1960, 1972; Zwislocki and Feldman, 1956; Sayers, 1964; Sayers and Cherry, 1957; Sayers and Lynn, 1968; Tobias and Zerlin, 1959; McFadden and Sharpley, 1972; Yost, 1974; Hornbostel and Wertheimer, 1920; von Bekesy, 1949, 1959, 1960; Blodgett et al., 1956; Pollack and Trittipoe, 1959a, b; Jeffress et al., 1962; Hirsh and Sherrick, 1961; Teas, 1962; Guttman, 1962; Harris, 1960; Harris et al., 1963; Babkoff and Sutton, 1966; Cherry and Taylor, 1954; Gulick, 1971; Pinheiro and Tobin, 1969; Flanagan et al., 1964; Moushegian and Jeffress, 1959; Whitworth and Jeffress, 1961;

Trimble, 1928; Stevens and Newman, 1936; Batteau et al., 1965; Batteau and Plante, 1962; Karlin, 1945; Sandel et al., 1955; Burger, 1958; Thurlow and Runge, 1967; Wallach, 1939, 1940; Wallach et al., 1949; Klensch, 1948; Steinberg and Snow, 1934; Hartley and Fry, 1921; Wightman and Firestone, 1930; Coleman, 1962, 1963)

4. How sensitive is the binaural auditory system to interaural time and amplitude differences?

(Durlach, 1972; Klumpp and Eady, 1956; Zwislocki and Feldman, 1956; Hershkowitz and Durlach, 1969a; Domnitz, 1973; Mills, 1960, 1972; Tobias and Zerlin, 1959)

5. How do interaural time and intensity differences "trade" in determining the lateralization of sounds?

Hornbostel and Wertheimer, 1920; Klemm, 1920; Shaxby and Gage, 1932; Hughes, 1940; Deatherage and Hirsh, 1959; David et al., 1959; Leakey, et al., 1958; Cherry and Sayers, 1959; Harris, 1960; Hafter and Carrier, 1972; Babkoff et al., 1973; Gilliom and Sorkin, 1972; Hershkowitz and Durlach, 1969b)

6. What are the limits of auditory frequency discrimination?

(Knudsen, 1923; Shower and Biddulph, 1931; Harris, 1952; Campbell and Small, 1963; Sekey, 1963; Plomp, 1964; Plomp and Bouman, 1959; Plomp and Levelt, 1965; Plomp and Mimpen, 1968; Nordmark, 1978; Terhardt, 1968; Soderquist, 1970; Fletcher, 1940; Hawkins and Stevens, 1950; Bilger and Hirsh, 1956; Shafer et al., 1950; Zwicker, 1952; Gassler, 1954; Zwicker and Feldtkeller, 1955; Bauch, 1956; Hamilton, 1957; Zwicker et al., 1957; Scharf, 1959, 1961; Greenwood, 1961; Green, 1965; Mulligan et al., 1968; Mulligan and Elrod, 1970; Henning, 1970; Thurlow and Rawlings, 1959; Thurlow and Bernstein, 1957; Pollack, 1964)

7. How is monaural intensity discrimination related to frequency and intensity of tonal inputs?

(Riesz, 1928; Miller, 1947; Tonndorf et al., 1955; Zwicker and Feldtkeller, 1967; Rabinowitz et al., 1976; McGill and Goldberg, 1968; Viemeister, 1972; Jesteadt et al., 1977; Fletcher and Munson, 1933; Robinson and Dadson, 1956)

8. What cues are available for monaural discrimination among the loudness of complex sounds?

(Miller, 1947 Hawkins and Stevens, 1950; Risset and Mathews, 1969; Grey and Moorer, 1977; Huggins, 1952; Miller and Carterette, 1975; Zwicker and Feldtkeller, 1955; Zwicker et al., 1957; Scharf, 1970, 1962, 1959; Pollack, 1952)

9. Are reaction times to acoustic stimuli dependent on loudness?

   (Chocholle, 1940, 1946, 1954; Chocholle and Greenbaum, 1966; Chocholle and DaCosta, 1971)

10. What are the joint effects of time uncertainty, S-R compatibility, and stimulus intensity on auditory choice-reaction times?

    (Sanders, 1977; Sanders and Wertheim, 1973; Bernstein et al., 1973)

11. How may stimulus-response compatibility be adjusted for optimum reaction times?

    (Smith, 1977; Hick, 1952; Hyman, 1953; Broadbent, 1971; Duncan, 1977; Broadbent and Gregory, 1965; Sanders, 1977; Greenwald, 1972; Greenwald and Shulman, 1973; Audley et al., 1975)

12. Is speed of reaction to stimuli influenced by the probability of stimulus occurrence?

    (Audley et al., 1975; Sanders, 1977; Naatanen and Merisalo, 1977; Yellott, 1971)

13. Is speed of vocal identification faster for geometric symbols than for numerals?

    (Forrin and Morin, 1967; Forrin, 1975)

14. Are reactions to stimuli of a symbolic nature (triangles, arrows, etc.) influenced by the order of presentation of such stimuli?

    (Falmagne et al., 1975)

15. Do choice-reaction times increase as the number of signals (choices) increase?

    (Hick, 1952; Hyman, 1953; Leonard, 1959; Smith, 1977; Sanders, 1977)

16. If an operator is prepared for rapid reaction to one stimulus, will his reaction time to a second stimulus be affected?

    (Audley, et al., 1975; Remington, 1969; Schvaneveldt and Chase, 1969)

17. What are the primary determinants of reaction times of successive responses to repeated events?

(Smith et al., 1973; Rabbitt et al., 1975; Rabbitt, 1966; Rabbitt and Vyas, 1973)

18. What task variables have been found to add or interact in their effects on reaction times?

(Sternberg, 1969; Rabbitt, 1967; Posner et al., 1973; Sanders, 1977; Sanders, 1970; Broadbent and Gregory, 1965; Brainard et al., 1962; Fitts et al., 1963; Miller and Pachella, 1973; Bertelson and Barzeele, 1965; Raab et al., 1961; Howell and Donaldson, 1962; Alegria and Bertelson, 1970; Egan et al., 1961)

19. Are reactions to auditory stimuli faster than reactions to visual stimuli?

(Trumbo and Gaillard, 1975; Kohfeld, 1971)

APPENDIX II.

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